Cisco Unified Communications Manager Express Command Reference

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accept

To allow a logical partitioning class of restriction (LPCOR) policy to accept calls associated with another resource-group, use the **accept** command in LPCOR policy configuration mode. To reject calls associated with a resource group, use the **no** form of this command.

accept *lpcor-group* [**fac**]
no accept *lpcor-group*

Syntax Description

lpcor-group	Name of the LPCOR resource group.
fac	Enables forced authorization code for calls from this resource group.

Command Default

Calls from other resource groups are rejected.

Command Modes

LPCOR policy configuration (cfg-lpcor-policy)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(3)T	Cisco Unified CME 8.5	This command was modified. The fac keyword was added to the accept command.

Usage Guidelines

Use this command to create a LPCOR policy by specifying the other resource groups from which this resource group can accept calls. If a resource group is not explicitly set to accept with this command, calls associated with that resource-group policy are rejected. You can create one LPCOR policy for each resource group.

If you create a LPCOR policy using the **voice lpcor policy** command and do not explicitly accept any other resource groups by using the **accept** command, that policy blocks all incoming calls associated with any LPCOR resource group other than its own. The fac keyword in the accept command restricts the caller from routing to a destination LPCOR group without entering a valid authorization code.

Examples

The following example shows the LPCOR policy for the resource group named sccp_phone_local. It accepts calls from the resource groups analog_phone_local and sip_phone_local but rejects calls from the group named analog_phone_remote because it is not included in the policy.

```
voice lpcor policy sccp_phone_local
 accept analog_phone_local
 accept sip_phone_local
```

The following example shows that sccp_phone_local blocks calls that are associated with any other LPCOR policy because its policy does not accept other resource groups.

```
voice lpcor policy sccp_phone_local
```

The following example shows that the policy local_phone is configured to not accept any calls associated with itself. SIP phone 1 and SCCP phone 2 both belong to the local_phone resource group and its policy prevents them from accepting calls from each other.

```
voice register pool  1
 lpcor type local
 lpcor incoming local_phone
 lpcor outgoing local_phone
 id mac 0021.A02D.B360
 type 7960
 number 1 dn 1
!
voice lpcor custom
 group 1 local_phone
 group 2 remote_phone
 group 3 analog_phone
!
voice lpcor policy local_phone
 no accept local_phone
 accept analog_phone
!
ephone  2
 lpcor type local
 lpcor incoming local_phone
 lpcor outgoing local_phone
 mac-address 0021.A02D.B580
 type 7960
 button  1:10
```

The following example shows that the authorization code is required by callers who belong to the LocalUser group and RemoteUser group.

```
!
voice lpcor policy PSTNTrunk
 service fac
 accept Manager
 accept LocalUser fac
 accept RemoteUser fac
 no accept PSTNTrunk
 no accept IPTrunk
```

Related Commands

Command	Description
show voice lpcor policy	Displays the LPCOR policy for the specified resource group.
voice lpcor custom	Defines the LPCOR resource groups on the Cisco Unified CME router.
voice lpcor policy	Creates a LPCOR policy for a resource group.

access-digit

To define the access digit that phone users dial to request a precedence call, use the **access-digit** command in voice MLPP configuration mode. To reset to the default, use the **no** form of this command.

access-digit *digit*
no access-digit

Syntax Description

digit	Single-digit number users dial. Range: 0 to 9. Default: 0.

Command Default

Access digit is 0.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Modification
12.4(22)YB	This command was introduced.
12.4(24)T	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command defines the MLPP access digit that a user must dial when making a precedence call. Phone users request a precedence call by dialing the prefix NP, where N is the preconfigured MLPP access digit and P is the requested precedence level, followed by the phone number.

Note Your domain type must support the access digit that you select. For example, the valid range for the DSN is 2 to 9.

Examples

The following example shows the MLPP access digit set to 6:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# access-digit 6
```

Related Commands

Command	Description
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.
preemption trunkgroup	Enables preemption capability on a trunk group.
preemption user	Enables preemption capability for all supported phones.

addons

To define the maximum number of add-on modules supported by the new Cisco Unified SIP IP phone on Cisco Unified CME, use the **addons** command in voice register pool mode. To remove the add-on modules , use the no form of this command.

addons *max-addons*
no addons *max-addons*

Syntax Description

max-addons	Defines the maximum number of addon modules that can be configured while defining the pool for the phone. Range is 1 to 3.

Command Default

The default value of the addons is 0. When the **reference-pooltype** command is configured, the add-on module value of the reference phone is inherited.

Command Modes

Voice Register Pool-type Configuration (config-register-pooltype)

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME. When you use the no form of this command, the inherited properties of the reference phone takes precedence over the default value.

Examples

The following example shows how to enter voice register pool configuration mode and define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME:

```
Router(config)# voice register pool 1
Router(config-register-pool-type)# type 9900 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool-type)# id mac 1234.4567.7891
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.
type	Defines a phone type for a SIP phone.

address (voice emergency response location)

To define the civic address for an ERL that is used for the ALI database upload, use the **address** command in voice emergency response location mode. To remove this definition, use the **no** form of the command. This command is optional.

address *string*
no address

Syntax Description

string	String (1-247 characters) used to identify an ERL's civic address.

Command Default

The civic address is not defined.

Command Modes

Voice emergency response location configuration (cfg-emrgncy-resp-location)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to create a comma separated text entry of the ERL's civic address. The address information must be entered to conform with the NENA-2 Data Record specifications or the recommendations by the service provider.

Examples

In this example, a civic address is displayed for ERL 60.

```
voice emergency response location 60
subnet 1 209.165.200.224 255.255.0.0
elin 1 4085550100
name Cookies and More Incorporated,
address I,408,5550100,,11902,,,Main Street,Emerald City,CA,Idina Menzel,1,,,,,,
```

Related Commands

Command	Description
elin	Specifies a PSTN number that will replace the caller's extension.
name	Specifies a string (up to 30 characters) used internally to identify or describe the emergency response location.
subnet	Defines which IP phones are part of this ERL.

addons

To define the maximum number of add-on modules supported by the new Cisco Unified SIP IP phone on Cisco Unified CME, use the **addons** command in voice register pool mode. To remove the add-on modules , use the no form of this command.

addons *max-addons*
no addons *max-addons*

Syntax Description

max-addons	Defines the maximum number of addon modules that can be configured while defining the pool for the phone. Range is 1 to 3.

Command Default

The default value of the addons is 0. When the **reference-pooltype** command is configured, the add-on module value of the reference phone is inherited.

Command Modes

Voice Register Pool-type Configuration (config-register-pooltype)

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME. When you use the no form of this command, the inherited properties of the reference phone takes precedence over the default value.

Examples

The following example shows how to enter voice register pool configuration mode and define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME:

```
Router(config)# voice register pool 1
Router(config-register-pool-type)# type 9900 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool-type)# id mac 1234.4567.7891
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.
type	Defines a phone type for a SIP phone.

after-hour exempt

To specify that an individual IP phone in Cisco Unified CME does not have any of its outgoing calls blocked even though after-hour call blocking has been enabled, use the **after-hour exempt** command in ephone or ephone-template configuration mode. To remove the exemption, use the **no** form of this command.

after-hour exempt
no after-hour exempt

Syntax Description

This command has no arguments or keywords.

Command Default

The SCCP phone is not exempt from call blocking.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in the ephone-template configuration mode was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

Use this command to exempt an individual SCCP phone from call blocking and enable the phone user to place outgoing calls regardless of whether the outgoing called number matches the defined pattern of digits during the call blocking periods.

By default, all IP phones in a Cisco Unified CME system are subject to call blocking if the Call Blocking feature is configured.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example shows how to configure this phone so that outgoing calls are not blocked:

```
Router(config)# ephone 23
Router(config-ephone)# mac 00e0.8646.9242
Router(config-ephone)# button 1:33
Router(config-ephone)# after-hour exempt
```

Related Commands

Command	Description
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.

Command	Description
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.

after-hour login http

To unblock an individual IP phone in Cisco Unified CME that is configured for after-hour call blocking, use the **after-hour login http** command in ephone, telephony-service or ephone-template configuration mode. To disable after-hour login http feature, use the no form of the command.

after-hour login htpp
no after-hour login htpp

Syntax Description

This command has no arguments or keywords.

Command Default

The after-hour login http feature is not enabled.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
T
elephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use this command to log in to a phone to unblock the after hour block and enable the phone user to place outgoing calls regardless of whether the outgoing called number matches the defined pattern of digits during the call blocking periods.

When you configure after-hours login http command, you will experience slightly different login behavior compare to the current one. This difference is because the after hours login mechanism is enhanced due to some UI limitation in the current model. By default, after-hours login http is not applied, which mean user will be using the existing after hours login mechanism.

By default, all IP phones in a Cisco Unified CME system are subject to call blocking if the Call Blocking feature is configured.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example shows how to configure this phone with pin login so that outgoing calls are not blocked:

```
Router(config)# ephone 6
Router(config-ephone)# mac 00e0.8646.242
Router(config-ephone)# button 1:33
Router(config-ephone)# Pin 123
Router(config-ephone)# after-hour login http
```

Related Commands

Command	Description
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.

after-hours block pattern

To define a pattern of outgoing digits for Call Blocking from IP phones, use the **after-hours block pattern** command in telephony-service or ephone-template configuration mode. To delete a call-blocking pattern, use the **no** form of this command.

after-hours block pattern *pattern-tag*
no after-hours block pattern *pattern-tag*

Syntax Description

pattern-tag	Identifier for a call-blocking pattern. Up to 100 call-blocking patterns can be defined in separate commands.
pattern	Outgoing call digits to be matched for blocking, including specific digit patterns and general regular expressions.
7-24	(Optional) If the **7-24** keyword is specified, the pattern is always blocked, 7 days a week, 24 hours a day. If the **7-24** keyword is not specified, the pattern is blocked during the days and dates that are defined with the **after-hours day** and **after-hours date** commands.

Command Default

No pattern is defined.

Command Modes

Ephone-template (config-ephone-template)

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4	Support for this command was extended to all SCCP, H.323, SIP, and POTS calls that go through the Cisco Unified CME router, including all incoming calls to the router, except calls from an exempt phone.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was added to ephone-template configuration mode.
12.4(15)XZ	Cisco Unified CME 4.3	This command was added to ephone-template configuration mode.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.3(2)T	Cisco Unified CME 9.5	This command was modified to include regular expressions as a value for the *pattern* argument.

Usage Guidelines

Call Blocking on IP phones is defined in the following way. First, one or more patterns of outgoing digits (0-9) are defined using the **after-hours block pattern** command. Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both. By default, all IP phones in a Cisco Unified CME system are restricted during the specified time if at least one pattern and at least one time period are defined.

Before Cisco CME 3.4, Call Blocking is supported on IP phones and on analog phones connected to SCCP-controlled analog telephone adaptors (Cisco ATA) or SCCP-controlled foreign exchange station (FXS) ports. In Cisco CME 3.4 and later, the call-blocking configuration applies to all SCCP, H.323, SIP and POTS calls that go through the Cisco Unified CME router. All incoming calls to the router, except calls from an exempt phone, are also checked against the after-hours configuration.

Individual phones can be exempted from call blocking using the **after-hour exempt**or the **after-hours override-code** command.

Blocked calls return a fast-busy tone to the user for approximately 10 seconds before the call is terminated and the line is returned to on-hook status.

In Cisco Unified CME 9.5 and Cisco Unified SRST 9.5, support for after-hours pattern blocking is extended to regular expression patterns for dial plans on Cisco Unified SIP and Cisco Unified SCCP IP phones. With this support, users can add a combination of fixed dial plans and regular expression-based dial plans.

When a call is initiated after hours, the dialed number is matched against a combination of dial plans. If a match is found, the call is blocked.

To enable regular expression patterns to be included when configuring after-hours pattern blocking, the **after-hours block pattern** command is modified to include regular expressions as a value for the *pattern* argument.

Note The maximum length of a regular expression pattern is 32 for both Cisco Unified SIP and Cisco Unified SCCP IP phones.

For a summary of the basic Cisco IOS regular expression characters and their functions, see the Cisco Regular Expression Pattern Matching Characters section of Terminal Services Configuration Guide.

Examples

The following example defines pattern 1, which blocks international calls after hours for a Cisco Unified CME system that requires dialing 9 for external calls:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 9011
```

The following example shows how to configure several after-hours block patterns of regular expressions:

```
Router# configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 ?
  WORD  Specific block pattern or a regular expression for after-hour block
        pattern
Router(config-telephony)# after-hours block pattern 1 1234
Router(config-telephony)# after-hours block pattern 2 .T
Router(config-telephony)# after-hours block pattern 3 987654([1-3])+
```

```
Router(config-telephony)# after-hours block pattern 4 98765432[1-9]
Router(config-telephony)# after-hours block pattern 5 98765(432|422|456)
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours override-code	Specifies that call blocking on an IP phone can be overridden by entering a defined code.
after-hours pstn-prefix	Specifies that trunk lines on an IP phone are blocked similarly to that configured for nonPSTN lines.
ephone-template (ephone)	Applies template to a SCCP phone.

after-hours date

To define a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones, use the **after-hours date** command in ephone-template or telephony-service configuration mode. To delete a defined time period, use the **no** form of this command.

after-hours date *month date start-time stop-time*
no after-hours date *month date*

Syntax Description

month	Abbreviated month. The following abbreviations for month are valid: **jan**, **feb**, **mar**, **apr**, **may**, **jun**, **jul**, **aug**, **sep**, **oct**, **nov**, **dec**.
date	Date of the month. Range is from 1 to 31.
start-time stop-time	Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time that is entered will be the next available time that follows the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.

Command Default

No time period based on date is defined for call blocking.

Command Modes

Ephone-template configuration (config-ephone-temp)

Telephony-service configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was added to ephone-template configuration mode.
12.4(15)XZ	Cisco Unified CME 4.3	This command was added to ephone-template configuration mode.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to define call blocking that recurs annually on the date specified in the command. Call blocking on IP phones is defined as follows:

- First, one or more patterns of outgoing digits (0-9) are defined using the **after-hours block pattern** command.

- Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both.

By default, all IP phones in a Cisco Unified CME system are restricted during the specified time if at least one pattern and at least one time period are defined. Individual IP phones can be exempted from call blocking using the **after-hour exempt or after-hours override-code** commands.

Examples

The following example defines January 1 as an entire day on which calls that match the pattern specified in the **after-hours block pattern** command are blocked:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours date jan 1 00:00 00:00
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.
after-hours block pattern	Defines a pattern of digits (0-9) for blocking outgoing calls from IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours override-code	Specifies that call blocking on an IP phone can be overridden by entering a defined set of digits (0-9).
after-hours pstn-prefix	Specifies that trunk lines on an IP phone are blocked similarly to that configured for nonPSTN lines.
ephone-template (ephone)	Applies template to SCCP phone.

after-hours day

To define a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones, use the **after-hours day** command in ephone-template or telephony-service configuration mode. To delete a defined time period, use the **no** form of this command.

after-hours day *day start-time stop-time*
no after-hours day *day*

Syntax Description

day	Abbreviated day of the week. The following abbreviations for day of the week are valid: **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, **sat**.
start-time *stop-time*	Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time that is entered will be the next available time that follows the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified day.

Command Default

No time period based on day of the week is defined for call blocking.

Command Modes

Ephone-template configuration (config-ephone-template)

Telephony-service configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was added to ephone-template configuration mode.
12.4(15)XZ	Cisco Unified CME 4.3	This command was added to ephone-template configuration mode.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to define call blocking during the hours between the start time and stop time on the day of the week that is specified in this command. This time period recurs weekly unless it is removed using the **no** form of this command.

Call blocking on IP phones is defined as follows:

- First, one or more patterns of outgoing digits (0-9) are defined using the **after-hours block pattern** command.

- Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both.

By default, all IP phones in a Cisco Unified CME system are restricted during the specified time if at least one pattern and at least one time period are defined. Individual phones can be exempted from call blocking using the **after-hour exempt** or **after-hours override-code** commands.

Examples

The following example defines the period from Monday night at 7 p.m. to Tuesday morning at 7 a.m. as an after-hours call-blocking period:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours day mon 19:00 07:00
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.
after-hours block pattern	Defines a pattern of digits (0-9) for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours override-code	Specifies that call blocking on an IP phone can be overridden by entering a defined set of digits (0-9).
after-hours pstn-prefix	Specifies that trunk lines on an IP phone are blocked similarly to that configured for nonPSTN lines.
ephone-template (ephone)	Applies template to SCCP phone.

after-hours override-code

To specify that a defined blocking pattern can be overridden, use the **after-hours override-code** command in ephone-template or telephony-service configuration mode. To remove the exemption, use the **no** form of this command.

after-hours override-code *pattern*
no after-hours override-code *pattern*

Syntax Description

pattern	Specifies the pattern of digits (0-9) that must be dialed by the phone user to override the call blocking rules. The override code is provided to the phone user by the system administrator.

Command Default

No override is defined.

Command Modes

Ephone-template configuration (config-ephone-template)

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was added to ephone-template configuration mode.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to allow a phone user to override call blocking rules and enable the phone user to place outgoing calls regardless of whether the outgoing called number matches the defined pattern of digits during the call blocking periods.

By default, all IP phones in a Cisco Unified CME system are subject to call blocking if the **Call Blocking** feature is configured. By entering the override code as defined by the system administrator, the phone user can override all call blocking rules.

The **after-hours override-code** command, configured by either ephone-template or telephony-service, overrides any global telephony-service call block configuration. If the **after-hour exempt** command is configured, it has priority over the **after-hours override-code** command.

Examples

The following example defines a blocking pattern using telephony-service configuration which can be overridden using the override code of **1234**:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 91900
Router(config-telephony)# after-hours day mon 19:00 07:00
Router(config-telephony)# after-hours date Jan 25 00:00 07:00
Router(config-telephony)# after-hours override-code 1234
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.
after-hours block pattern	Defines a pattern of digits (0-9) for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours pstn-prefix	Specifies that trunk lines on an IP phone are blocked similarly to that configured for non PSTN lines.
ephone-template (ephone)	Applies a template to an ephone.

after-hours pstn-prefix

To specify that all configured blocking patterns apply to trunk or PSTN lines, use the **after-hours pstn-prefix** command in telephony-service configuration mode. To delete call blocking configuration for PSTN lines, use the **no** form of this command.

after-hours pstn-prefix *tag pattern*
no after-hours pstn-prefix *tag pattern*

Syntax Description

tag	Identifier for a PSTN call-blocking pattern. Up to 4 call-blocking patterns can be defined in separate commands.
pattern	Outgoing call digits (0-9) to be matched for PSTN blocking.

Command Default

No pattern is defined.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was added to ephone-template configuration mode.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use the **after-hours pstn-prefix** command to indicate that the after-hours call blocking patterns are configured to include one or more PSTN access digits that are normally dialed by phone users using regular ephone-dn lines. For example, the patterns are configured with a leading digit 9 to correspond to the conventional “dial 9 for outside line.” The **after-hours pstn-prefix** command instructs the system to skip over the PSTN prefix digits (in the blocking patterns) for calls that are dialed directly to the PSTN on ephone-dns that are configured using the trunk feature. These lines do not require the user to dial a PSTN access code (for example, 9) because they are configured to directly connect to the PSTN FXO ports. For example, a user of a regular ephone-dn would dial 9-1-900-xxx-xxxx, whereas a user on a trunk ephone-dn would omit the leading 9 and dial only 1-900-xxx-xxxx. The blocking pattern would be configured as **91900** to restrict calls on regular ephone-dn lines, and this pattern would be interpreted as 1900 on ephone-dns configured using the trunk feature. If you do not specify the **after-hours pstn-prefix** command, then no blocking is performed on calls dialed on trunk ephone-dn lines.

Call blocking on IP phones is defined as follows:

- First, one or more patterns of outgoing digits (0-9) are defined using the **after-hours block pattern** command.
- Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the **after-hours date,** the **after-hours day,** or both commands.

By default, all IP phones in a Cisco Unified CME system are restricted during the specified time if at least one pattern and at least one time period are defined.

Blocked calls return a fast-busy tone to the user for approximately 10 seconds before the call is terminated and the line is returned to on-hook status.

Examples

The following example defines a blocking pattern using telephony-service configuration which is applied to a PSTN line:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 91900
Router(config-telephony)# after-hours pstn-prefix 1 9
Router(config-telephony)# after-hours day mon 19:00 07:00
Router(config-telephony)# after-hours date Jan 25 00:00 07:00
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.
after-hours block pattern	Defines a pattern of digits (0-9) for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours override-code	Specifies that call blocking on an IP phone can be overridden by entering a defined series of digits (0-9).

allow watch

To allow a directory number on a phone registered to Cisco Unified CME to be watched in a presence service, use the **allow watch** command in ephone-dn, ephone-dn-template, or voice register dn configuration mode. To reset to the default condition, use the **no** form of this command.

allow watch
no allow watch

Syntax Description

This command has no arguments or keywords.

Command Default

Watching of the phone line is disabled.

Command Modes

Ephone-dn configuration (config-ephone)
Ephone-dn-template configuration (config-ephone-dn-template)
Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command controls whether a phone line associated with a directory number can be watched as part of a presence service. The directory number is enabled as a presentity that can be watched by internal and external watchers. Presence service must be enabled on Cisco Unified CME. Another phone, acting as a watcher, can monitor the status of this phone line when the **blf-speed-dial** or **presence call-list** command is enabled for that phone.

If you use an ephone-dn template to apply a command to a directory number and you also use the same command in ephone-dn configuration mode, the value that you set in ephone-dn configuration mode has priority over the ephone-dn template configuration.

Examples

The following example shows that the extension associated with voice register dn 2 can be watched by the phone associated with voice register pool 1:

```
Router(config)# voice register dn 2
Router(config-register-dn)# number 2102
Router(config-register-dn)# allow watch
Router(config)# voice register pool 1
Router(config-register-pool)# id mac 0015.6247.EF90
Router(config-register-pool)# type 7971
Router(config-register-pool)# number 1 dn 2
Router(config-register-pool)# blf-speed-dial 1 2102 label 2102
```

Related Commands

Command	Description
blf-speed-dial	Enables Busy Lamp Field (BLF) monitoring for a speed-dial number on a phone registered to Cisco Unified CME.
presence	Enables presence service and enters presence configuration mode.
presence call-list	Enables BLF monitoring for call lists and directories on phones registered to Cisco Unified CME.
presence enable	Allows the router to accept incoming presence requests.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.

anonymous block

To enable anonymous call blocking in a SIP phone template, use the **anonymous block** command in voice register template configuration mode. To return to the default, use the **no** form of this command.

anonymous block
no anonymous block

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command blocks incoming calls in which the caller is not identified. To apply a template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following example shows how to set anonymous call blocking in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# anonymous block
```

Related Commands

Command	Description
caller-id block (voice register template)	Enables caller-ID blocking for outbound calls from a SIP phone.
template (voice register pool)	Applies a template to a SIP phone.

application (telephony-service)

To select a session-level application for all extensions (ephone-dns) in a Cisco Unified CME, use the **application** command in telephony-service configuration mode. To disable use of an application for all extensions, use the **no** form of this command.

application *application-name*
no application

Syntax Description

application-name	Interactive voice response (IVR) application name.

Command Default

No application is selected for all extensions.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Use this command to assign a Tool Command Language (Tcl) IVR application to all extensions served by the CME router.

Use the **show call application voice summary** command to display a list of applications.

Examples

The following example sets the IVR application for all phones:

```
Router(config)# telephony-service
Router(config-telephony) application TCL IVR
```

Related Commands

Command	Description
show call application voice summary	Displays information about voice applications.

application (voice register global)

To select the session-level application for all dial peers associated with Session Initiation Protocol (SIP) phones, use the **application** command in voice register global configuration mode. To disable use of the application, use the **no** form of this command.

application *application-name*
no application

Syntax Description

application-name	Interactive voice response (IVR) application name.

Command Default

Default application on router

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

During Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) registration, a dial peer is created for each SIP phone and that dial peer includes the default session application. The **application** command allows you to change the default application for all dial peers associated with the Cisco SIP IP phones, if desired. The applied application (or TCL IVR script) must support call redirection. Use the **show call application voice summary** command to display a list of applications.

The **application** command in voice register pool configuration mode takes precedence over this command in voice register global configuration mode.

Note Configure the **id** (voice register pool) command before any other voice register pool commands, including the **application** command. The **id** command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

Examples

The following example shows how to set the Tcl IVR application globally for all SIP phones:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# application sipapp2
```

Related Commands

Command	Description
application (dial-peer)	Enables a specific application on a dial peer.

Command	Description
application (voice register pool)	Selects the session-level application for the dial peer associated an individual SIP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
show call application voice summary	Displays information about voice applications.
show dial-peer voice	Displays information for dial peers.
voice register pool	Enters voice register pool configuration mode for SIP phones.

application (voice register pool)

To select the session-level application for the dial peer associated with an individual Session Initiation Protocol (SIP) phone in a Cisco Unified CallManager Express (Cisco Unified CME) environment or for a group of phones in a Cisco Unified SIP Survivable Remote Site Telephony (SRST) environment, use the **application** command in voice register pool configuration mode. To disable use of the application, use the **no** form of this command.

application *application-name*
no application

Syntax Description

application-name	Name of the selected interactive voice response (IVR) application name.

Command Default

Default application on router

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.

Usage Guidelines

During Cisco Unified CME or Cisco Unified SIP SRST registration, a dial peer is created for each SIP phone and that dial peer includes the default session application. The **application** command allows you to change the default application for all dial peers associated with the Cisco SIP IP phones, if desired. The applied application (or TCL IVR script) must support call redirection. Use the **show call application voice summary** command to display a list of applications.

The **application** command in voice register pool configuration mode takes precedence over this command in voice register global configuration mode.

Note Configure the **id** (voice register pool) command before any other voice register pool commands, including the **application** command. The **id** command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

Examples

The following example shows how to set the IVR application for the SIP phone specified by the **voice register pool** command:

```
Router(config)# voice register pool 1
Router(config-register-pool) application sipapp2
```

The following partial sample output from the **show running-config** command shows that voice register pool 1 has been set up to use the SIP.app application:

```
voice register pool 1
 id network 172.16.0.0 mask 255.255.0.0
 application SIP.app
 voice-class codec 1
```

Related Commands

Command	Description
application (dial-peer)	Enables a specific application on a dial peer.
application (voice register global)	Selects the session-level application for all dial peers associated with SIP phones.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
show call application voice summary	Displays information about voice applications.
show dial-peer voice	Displays information for dial peers.

apply-config

To dynamically apply the phone configuration on Cisco Unified SIP IP phones 8961, 9951, and 9971, without restarting the phones, use the **apply-config** command in voice register global and voice register pool configuration modes.

apply-config

Syntax Description

This command has no arguments or keywords.

Command Default

Apply-config is not enabled by default.

Command Modes

Voice register global
Voice register pool

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.

Usage Guidelines

Use this command to dynamically apply the phone configuration on Cisco Unified SIP IP phones 8961, 9951, and 9971.Once you configure the apply-config command, you are not required to restart the phone. The phone restarts by itself or dynamically applies the changes to the phone configuration without restarting.

Examples

The following example shows the apply-config command configured in voice register pool 5 :

Router# configure terminal

Router(config)#voice register pool 5

Router(config-register-pool)#apply-config

Related Commands

Command	Description
camera	Enables USB camera capability on Cisco Unified IP Phones 9951 and 9971
video	Enables video capability on Cisco Unified SIP IP Phones 9951 and 9971

ata-ivr-pwd

To define a password to access interactive voice response (IVR) and change the default phone settings on Cisco Analog Telephone Adaptors, use the **ata-ivr-pwd** command in voice register pool configuration mode. To return to the default, use the **no** form of the command.

ata-ivr-pwd **[0|6]** *password*
no **ata-ivr-pwd**

Syntax Description

password	Four-digit string to be used as password to access IVR. Password string must contain numbers 0 to 9. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Command Default

No valid password is set.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
15.2(2)T	Unified CME 9.0	This command was introduced.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption. Password policy is not enforced, as the command supports a four-digit password.

Examples

The following example shows how 1234 is defined as the password to access IVR on Cisco ATA-187:

```
voice service voip
 allow-connections sip to sip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
voice register pool  11
 ata-ivr-pwd 1234
 id mac 93FE.12D8.2301
 session-transport tcp
 type ATA-187
number 1 dn 33
 username ata112 password cisco
 codec g711ulaw
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.

attempted-registrations size

To set the size of the table that shows a number of attempted-registrations, use the attempted- registrations command in voice register global mode. To set the size of attempted-registrations table to its default value, use the no form of this command.

attempted-registrations size size
no attempted-registrations size

Syntax Description

size	Number of entries in attempted registrations table. Size range from 0 to 50.

Command Default

The default size for attempted registration table is 10.

Command Modes

voice register global

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to define the size of the table that stores information related to SIP phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail. The default size of an attempted registration table is 10. The minimum size of attempted registration table is 0. Use the attempted-registration size 0 when you do not wish to store any information about phones attempting to register with the Cisco Unified CME or Cisco Unified SRST and fail. The maximum size of attempted registration table is 50.

When the current number of entries in the table is more than the new size that is being configured, system prompts the user for the following confirmation, “This will remove x old entries from the table. Proceed? Yes/No ?”. The default user confirmation is “No”. Where “x” represents the number of entries that will be deleted. The old entries are classified on basis of the time-stamp of the latest register attempt made by the phone.

During rollback, the user confirmation is not sought and the target configuration is applied. If the current number of entries in the table is more than the default value of the table size, then entries in excess of the default table size are cleared before reverting to the target table size.

For example, if the configured table size is 40 and there are currently 35 entries in the table, any change in the size of the attempted registration table during rollback removes 25 oldest entries leaving only the default (10) entries before making the table size equal to the size in target configuration.

Examples

The following example shows attempted-registrations size:

```
Router# conf t
Router(config)#voice register global
Router(config-register-global)#attempted-registrations size 15
!
```

Related Commands

Command	Description
clear voice register attempted- registrations	Allows to delete entries in attempted-registration table.
show voice register attempted-registrations	Displays details of phones that attempted to register and failed.

attendant-console

To specify the phone number of the MLPP attendant-console service, use the **attendant-console** command in voice MLPP configuration mode. To revert to the default, use the **no** form of this command.

attendant-console *number* **redirect-timer** *seconds*
no attendant-console

Syntax Description

number	Pilot number of the MLPP attendant-console service, such as the Cisco Unified CME basic automatic call distribution (B-ACD) and auto-attendant (AA) service.
seconds	Number of seconds that a call rings before being redirected to the attendant-console service. Range: 10 to 60.

Command Default

MLPP call is not diverted to an attendant-console service.

Command Modes

V
oice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	—	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command enables Cisco Unified CME to divert all unanswered precedence calls above Routine to the specified target number after the specified period of time. This target directory number typically specifies the pilot number of the attendant-console service that is used as a destination of last resort for forwarded MLPP calls.

Examples

The following example shows that any MLPP call that is not answered after 30 seconds is redirected to extension 81005, which is the extension of the BACD queue.

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# attendant-console 81005 redirect-timer 30
```

Related Commands

Command	Description
access-digit	Defines the access digit that phone users dial to request a precedence call.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.
service	Associates a dial peer with an auto-attendant (AA) service.

audible-tone

To configure audible tones to indicate successful join or unjoin and login or logout from any hunt group, use the **audible-tone** command in ephone or ephone-template configuration mode. To revert to the default behavior of not playing any audible tone, use the **no** form of this command.

audible-tone
no audible-tone

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this feature is disabled.

Command Modes

Ephone configuration (config-ephone)

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

Use the **audible-tone** command to set an audible tone to confirm successful join or unjoin and log in or log out from specific hunt groups.

Example

The following example shows how to configure audible tone in ephone configuration mode:

```
Router(config)# ephone 1
Router(config-ephone)# audible-tone
```

The following example shows how to configure an audible tone in ephone-template configuration mode:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# audible-tone
```

Related Commands

Command	Description
ephone-hunt group	Configures an ephone hunt group in Cisco Unified CME.
voice-hunt group	Configures a voice hunt group in Cisco Unified CME.

authen-method

To define authentication method for a vpn-profile, use the **authen-method** command in vpn-profile configuration mode. To disable the authentication method, use the no form of this command.

authen-method [{*bothnonepassword*}]
no authen-method

Syntax Description

both	Requires both user id and password to authenticate.
password	Requires only password to authenticate.
none	Does not allows authentication.

Command Default

Both User ID and Password are required for authentication.

Command Modes

Voice service voip (cfg-lpcor-policy)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to define authentication method for a vpn-profile. You can define an authen-method with both user id and password, or you can define an authen-method with just password. You can choose to not allow any authentication method by configuring authen-method none.

Examples

The following example shows the authen-method both defined for vpn-profile 2:

```
Router# show run
!
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  keepalive 50
  auto-network-detect enable
  host-id-check disable
 vpn-profile 2
  mtu 1300
  authen-method both
  password-persistent enable
  host-id-check enable
 vpn-profile 4
  fail-connect-time 50
```

Related Commands

Command	Description
vpn-profile	Defines a VPN-profile.

authenticate (voice register global)

To define the authenticate mode for SIP phones in a Cisco Unified CME or Cisco Unified SRST system, use the **authenticate** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

Cisco IOS Release 12.4(11)XJ and Later Releases

authenticate {**credential** *tag location* | **ood-refer** | **presence** | **realm** *string* | **register**}

no authenticate {**credential** *tag location* | **ood-refer** | **presence** | **realm** *string* | **register**}

Cisco IOS Release 12.4(4)T

authenticate [**all**] [**realm** *string*]

no authenticate [**all**] [**realm** *string*]

Syntax Description

credential *tag*	Number that identifies the credential file to use for out-of-dialog REFER (OOD-R) or presence authentication. Range: 1 to 5.
location	Name and location of the credential file in URL format. Valid storage locations are TFTP, HTTP, and flash memory.
ood-refer	Incoming OOD-R requests are authenticated using RFC 2617-based digest authentication.
presence	Incoming presence subscription requests from an external presence server are authenticated.
realm *string*	Realm parameter for challenge and response as specified in RFC 2617 is authenticated.
register	All incoming registration requests are challenged and authenticated. Valid for Cisco Unified CME only.

Command Default

Authenticate mode is disabled.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	The **credential**, **ood-refer**, **presence**, and **register** keywords were added. The **register** keyword replaced the **all** keyword.
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	The modifications to this command were integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

The **credential** keyword allows OOD-R and presence service to use credential files for authentication. Up to five text files containing username and password pairs can be defined and loaded into the system. The contents of these five files are mutually exclusive; the username and password pairs must be unique across all the files. For Cisco Unified CME, the username and password pairs cannot be the same ones defined for SCCP or SIP phones with the **username** command.

The **ood-refer** keyword specifies that any OOD-R request that passes authentication is authorized to setup calls between referee and refer-target if OOD-R is enabled with the **refer-ood enable** command.

The **presence** keyword enables digest authentication for external watchers. Credentials are verified against a credential file stored in flash. This applies to both OOD-R and presence. The default is to authenticate all SUBSCRIBE requests from external watchers. An external watcher that passes authentication is authorized to subscribe to presence service for all lines allowed to be watched.

The **register** keyword enables authentication for registration requests in which the MAC address of the SIP phone cannot be identified by using other methods. All incoming register requests are challenged and authenticated. The **realm** keyword with the *string* argument specifies the character string to be included in the challenge.

Examples

The following example shows that all registration requests from SIP phones in a Cisco Unified CME system must be authenticated:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# authenticate register
```

Related Commands

Command	Description
credential load	Reloads a credential file into flash memory.
mode cme	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
presence-enable	Allows incoming presence subscribe requests from SIP trunks.
refer-ood enable	Enables OOD-R processing.
username (ephone)	Defines a username and password for SCCP phones.
username (voice register pool)	Defines a username and password for authenticating SIP phones.

authentication credential

To create an entry for an application's credential in the database used by the Cisco Unified CME authentication server, use the **authentication credential** command in telephony-service configuration mode. To remove the credential, use the **no** form of this command.

authentication credential *application-name password*
no authentication credential *application-name password*

Syntax Description

application-name	String sent by application to identify itself to the server. Length of string: 1 to 15characters.
password	String sent by application to identify itself to the server. Length of string: 1 to 15 characters. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Command Default

The credential is not stored in the database.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption, based on Unified CME password policy.

Usage Guidelines

This command stores a credential in the database used by the Cisco Unified CME authentication server. The authentication server uses this data to authenticate and authorize HTTP requests from IP phones in Cisco Unified CME.

Up to eight credentials can be stored in the database for the Cisco Unified CME authentication server.

For applications other than Extension Mobility, the credential must be created in the application.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]**. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Note This command is not required for authorizing requests from Extension Mobility phones in Cisco Unified CME.

Examples

The following example shows how to configure IP phones in Cisco Unified CME to request authorization from the internal authentication server. When the IP phone receives a command from the application, the phone requests authorization from the Cisco Unified CME authentication server to execute that command. The authorization request from the phone includes the specified credential. The authentication server compares the credential in its database to the one in the request, and authorizes or rejects the request based on the results.

```
Router(config)# telephony-service
Router(config-telephony)# authentication credential att psswrd
Router(config-telephony)# url authentication http://192.0.2.0/CCMCIP/authenticate.asp att
psswrd
Router(config-telephony)#
```

Related Commands

Command	Description
url authentication	Specifies authentication server and credential to be used by an application.

auto assign

To automatically assign an already defined telephone or extension number to button 1 of Cisco Unified IP phones as they register for service with a Cisco Unified CME router, use the **auto assign command in** telephony-service configuration mode. To return to the default of not automatically assigning dn-tags, use the **no** form of this command.

auto assign *dn-tag* **to** *dn-tag* [**type** *phone-type*] [**cfw** *extension-number* **timeout** *seconds*]
no auto assign *dn-tag* **to** *dn-tag* [**type** *phone-type*] [**cfw** *extension-number* **timeout** *seconds*]

Syntax Description

dn-tag **to** *dn-tag*	Range of ephone-dn tags for already configured ephone-dns, from which a tag is assigned to the ephone being created. The maximum number of directory numbers supported is version and platform dependent. Type **?** to display the value.
type *phone-type*	(Optional) Type of Cisco Unified IP phone to which to restrict automatic assignment of ephone-dn tags. Valid entries are the following: • **12SP**—12SP+ and 30VIP phones. • **7902**—Cisco Unified IP Phone 7902G. • **7905—**Cisco Unified IP Phone 7905G. • **7906—**Cisco Unified IP Phone 7906G. • **7910—**Cisco Unified IP Phone 7910 and 7910G. • **7911**—Cisco Unified IP Phone 7911G. • **7912—**Cisco Unified IP Phone 7912G. • **7920**—Cisco Unified Wireless IP Phone 7920. • **7921**—Cisco Unified Wireless IP Phone 7921. • **7931**—Cisco Unified Wireless IP Phone 7931G. • **7935—**Cisco Unified IP Conference Station 7935. • **7936—**Cisco Unified IP Conference Station 7936. • **7937**—Cisco Unified IP Conference Station 7937 • **7940—**Cisco Unified IP Phones 7940 and 7940G. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941G-GE. • **7942**—Cisco Unified IP Phone 7942. • **7945**—Cisco Unified IP Phone 7945.

type *phone-type*	• **7960—**Cisco Unified IP Phones 7960 and 7960G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961G-GE. • **7962**—Cisco Unified IP Phone 7962. • **7965**—Cisco Unified IP Phone 7965. • **7970**—Cisco Unified IP Phone 7970G. • **7971**—Cisco Unified IP Phone 7971G-GE. • **7975**—Cisco Unified IP Phone 7975. • **7985**—Cisco Unified IP Phone 7985. • **CIPC**—Cisco IP Communicator. • **all**—All ephone types. • **anl**—Analog gateway. • **ata—**Cisco ATA-186 or Cisco ATA-188. • **bri**—SCCP gateway. • **vgc-phone**—vg248 phone emulation for analog phone. **Note** You can also add a new phone type to your configuration by using the **ephone-type** command.
cfw	(Optional) Automatically assigned ephone-dns are provisioned for call-forward busy and no-answer to the specified extension number.
extension-number	(Optional) Extension number to which calls are to be forwarded on busy and no-answer conditions.
timeout *seconds*	(Optional; required if the **cfw** keyword is used) Amount of time, in seconds, to wait when a call is not being answered before forwarding it. Range: 3 to 60000.

Command Default

Ephone-dn tags are not automatically assigned to registering Cisco Unified IP phones.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2.1	The **7970** keyword was added.
12.3(14)T	Cisco CME 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The **7941**, **7941GE**, **7961**, and **7961GE** keywords were added.

Cisco IOS Release	Cisco Product	Modification
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	The **7931** keyword was added.
12.4(4)XC4	Cisco Unified CME 4.0(3)	The **7931** keyword was added.
12.4(11)T	Cisco Unified CME 4.0(3)	This command was integrated into Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **7921** and **7985** keywords were introduced.
12.4(15)T	Cisco Unified CME 4.1	This command wasintegrated into Cisco IOS Release 12.4(15)T.
12.4(15)T1	Cisco Unified CME 4.1(1)	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were introduced.
12.4(11)XW3	Cisco Unified CME 4.2	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were introduced.
12.4(15)XZ	Cisco Unified CME 4.3	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were introduced.
12.4(20)T	Cisco Unified CME 7.0	The **7937** keyword was introduced and this command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to create an ephone configuration for a Cisco Unified IP phone whose MAC address is not explicitly configured as it registers in Cisco Unified CME. The system-created ephone configuration includes the MAC address of the Cisco Unified IP phone being registered and an already-defined available ephone-dn assigned to button 1 of this phone.

The **auto-reg-ephone** command must be enabled (default) to use this command. If the auto registration feature is disabled, a Cisco Unified IP phone whose MAC address is not explicitly configured cannot register in Cisco Unified CME.

Before using this command, configure the ephone-dn tags to be assigned and define at least one primary number for each dn-tag.

All ephone-dns in a specified range should be of the same type, either single-line or dual-line.

Ephone-dn tags to be assigned must belong to normal ephone-dns and cannot belong to paging ephone-dns, intercom ephone-dns, music-on-hold (MOH) ephone-dns, or message-waiting-indication (MWI) ephone-dns.

The **auto assign** command cannot create shared lines.

If an insufficient number of dn-tags is available, some ephone configurations will not include a telephone or extension number.

Use multiple **auto assign** commands to assign discontinuous ranges of ephone-dn tags and to support multiple types of IP phones. Overlapping ranges of dn-tags may be assigned so that they map to more than one type of phone. If no **type** is specified, the values in the range are assigned to phones of any type, and if a specific range is assigned for a specific phone type, the available ephone-dn tag in that range are used first.

If the phone being registered is connected to a Cisco VG200 series analog phone gateway, configuring the

auto assign command will automatically create one ephone configuration for each configured port, as the port registers with the Cisco Unified CME router. To ensure that the tag-to-port assignment will match the numbering order of the physical ports; for example, dn-tags 1 to 24 assigned to ports 1 to 24 of a Cisco VG224 analog phone gateway, in that order, we recommend that the Cisco Unified CME system be up, running, and configured *before* you boot the analog phone gateway.

The **auto assign** command cannot be used for the Cisco Unified IP Phone 7914 Expansion Module. Phones with one or more expansion modules must be configured manually.

After using this command, reboot the phone for which an ephone is to be configured.

This command is also used by the Cisco Unified CME setup tool to automatically assign ephone-dns after the tool has gathered information about the setup from the user. When lines are assigned by the Cisco Unified CME setup tool in keyswitch mode with two ephone-dn entries created for each individual extension number, the automatic assignment mechanism assigns both ephone-dn entries to an individual ephone associated with an IP phone.

Note Care should be taken when using the **auto assign** command because this command grants telephony service to *any* IP phone that attempts to register. If you use the **auto assign** command, ensure that your network is secure from unauthorized access by unknown IP phones.

Examples

The following examples show how to configure the Auto Assign feature, including prerequisite commands for configuring the **auto assign** command.

The following example shows how to enter the ephone-dn configuration and create ephone-dns configurations, tags 1-4, each having a single primary number:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2000
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 3000
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 3
Router(config-ephone-dn)# number 4000
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 4001
Router(config-ephone-dn)# exit
```

The following example shows how to designate ephone-dn tags 1 to 4 for automatic assignment to any type of IP phone and then perform a fast reboot of all phones:

```
Router(config)# telephony-service
Router(config-telephony)# auto assign 1 to 4
Router (config-telephony)# restart all
```

The following example is the partial output from the **show ephone registered** command listing four registered IP phones, to which ephone-dn tags 1 to 4 have been automatically assigned, after the phones were booted:

```
Router# show ephone registered
ephone-1 Mac:0003.E3E7.F627 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.2 51671 Telecaster 7940 keepalive 28 max_line 2
button 1: dn 1 number 2000
ephone-2 Mac:0030.94C3.F43A TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.3 50094 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 2 number 3000
ephone-3 Mac:0003.6B40.99DA TCP socket:[3] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.200 51879 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 3 number 4000
ephone-4 Mac:0010.406B.99D9 TCP socket:[4] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.012 51879 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 4 number 4001
.
.
.
```

The following example shows how to designate ephone-dn tags 1 to 12 for automatic assignment to Cisco Unified IP Phone 7910Gs only and ephone-dn tags 13 to 20 for automatic assignment to a Cisco Unified IP Phones 7960 and 7960Gs only, with call forwarding to extension 5001 on busy or after 30 seconds of ringing with no answer:

```
Router(config)# telephony-service
Router(config-telephony)# auto assign 1 to 12 type 7910
Router(config-telephony)# auto assign 13 to 20 type 7960 cfw 5001 timeout 30
```

Related Commands

Command	Description
auto-reg-ephone	Enables registration of Cisco Unified IP phones for which MAC addresses are not explicitly configured.
number	Associates a telephone or extension number with an ephone-dn.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco Unified CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show ephone	Displays statistical information about registered Cisco Unified IP phones.
show ephone registered	Displays the status of registered phones.

auto-assign (auto-register)

To configure the mandatory DN range for automatic registration of SIP phones with the Cisco Unified CME system, use the **auto-assign** command in voice auto register configuration mode. This command is a sub-mode CLI of the command **auto-register** . To disable configuring DN range for auto registration of SIP phones, use the **no** form of this command.

auto-assign *First DN number* **to** *Last DN number*
no auto-assign

Syntax Description

auto-assign *First DN number* **to** *Last DN number*	The mandatory range of directory numbers configured for phones auto registering on Unified CME. Range: 1 to 4294967295.

Command Default

By default, this command is disabled.

Command Modes

voice auto register configuration (config-voice-auto-register)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

This command is a sub-mode option of the command **auto-register**. The command enables the administrator to configure the DN range for the SIP phones auto registering on Unified CME. For SIP phones to successfully auto register on Unified CME, it is mandatory that the DN range is defined. The assigned value of First DN number should be greater than zero.

Examples

The following example shows how to configure DN range for auto registration of SIP phones:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)# ?

VOICE auto register configuration commands:
auto-assign Define DN range for auto assignment
default Set a command to its defaults
exit Exit from voice register group configuration mode
no Negate a command or set its defaults
password Default password for auto-register phones
service-enable Enable SIP phone Auto-Registration
template Default template for auto-register phones

Router(config-voice-auto-register)#auto-assign ?
  <1-4294967295> First DN number
Router(config-voice-auto-register)#auto-assign 1 ?
   to to
Router(config-voice-auto-register)#auto-assign 1 to ?
  <1-4294967295> Last DN number
Router(config-voice-auto-register)#auto-assign 1 to 10
```

Related Commands

Command	Description
service-enable (auto-register)	Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.
password (auto-register)	Configures the mandatory password that administrator sets for auto registration of SIP phones on Unified CME.
auto-register	Enables automatic registration of SIP phones with the Cisco Unified CME system.
template (auto-register)	Creates a basic configuration template that supports all the configurations available on the voice register template.
auto-reg-ephone	Enables automatic registration of ephones with the Cisco Unified CME system.

auto logout

To enable the automatic change of an ephone hunt group agent's ephone-dn to not-ready status after a specified number of hunt-group calls are not answered, use the **auto logout** command in ephone-hunt configuration mode. To disable automatic logout, use the **no** form of this command.

auto logout [*number-of-calls*] [{**dynamic** | **static**}]
no auto logout [*number-of-calls*] [{**dynamic** | **static**}]

Syntax Description

number-of-calls	(Optional) Number of unanswered hunt-group calls to an ephone-dn before the ephone-dn is automatically changed to not-ready status. Range is from 1 to 20. Default is 1.
dynamic	(Optional) Specifies that this command applies only to dynamic hunt group members (those who are specified by an asterisk (*) wildcard in the hunt group configuration). If neither the **dynamic** nor **static** keyword is used, automatic logout applies to both dynamic and static hunt group members.
static	(Optional) Specifies that this command applies only to static hunt group members (those whose extension numbers are explicitly named in the hunt group configuration). If neither the **dynamic** nor **static** keyword is used, automatic logout applies to both dynamic and static hunt group members.

Command Default

Automatic change of agent status to not-ready is disabled.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The *number-of-calls* argument and the **dynamic** and **static** keywords were added. The criterion for this command was changed from exceeding the **timeout** command limit to exceeding the number of calls specified in this command.
12.4(9)T	Cisco Unified CME 4.0	The modifications made to this command were integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

This command is valid only for the following Cisco IP phones:

- Cisco Unified IP Phone 7905G
- Cisco Unified IP Phone 7912G
- Cisco Unified IP Phones 7940 and 7940G
- Cisco Unified IP Phones 7960 and 7960G

This command is used with the Automatic Agent Status Not-Ready feature for ephone hunt groups, which automatically puts an agent's phone in not-ready status when it exceeds a specified limit. The limit at which the Automatic Agent Status Not-Ready feature is triggered depends on the Cisco CME version that you are using, as follows:

- Cisco CME 3.3 and earlier versions—Automatic Agent Status Not-Ready is invoked when an ephone-hunt group call rings longer on a member ephone-dn than the period of time configured in the **timeout** command in ephone-hunt configuration mode.
- Cisco Unified CME 4.0 and later versions—Automatic Agent Status Not-Ready is invoked when the specified number of ephone-hunt group calls is unanswered by an agent. The default is one call if the number of calls is not explicitly specified.

When Automatic Agent Status Not-Ready is specified for an ephone hunt group and it is triggered because an ephone-dn member does not answer a specified number of ephone hunt group calls, the following actions take place:

- If the **hunt-group logout HLog** command has been used, the agent is placed in not-ready status. The agent's phone will not receive further hunt-group calls but will receive calls that directly dial the phone's extension numbers. An agent in not-ready status can return to ready status by pressing the HLog soft key or by using the HLog feature access code (FAC).
- If the **hunt-group logout HLog** command has not been used or if the **hunt-group logout DND** command has been used, the phone on which the ephone-dn appears is placed into Do Not Disturb (DND) mode, in which the ephone-dn does not receive any calls at all, including hunt-group calls. The red lamp on the phone lights to indicate DND status. An agent in DND mode can return to ready status by pressing the DND soft key or by using the DND FAC.
- When an agent returns to ready status, the ephone hunt group resumes sending calls to the agent's ephone-dn.

Note When an agent who is a dynamic member of a hunt group is in not-ready status, the agent's slot in the ephone hunt group is not relinquished. It remains reserved by the agent until the agent leaves the group.

You can use the **auto logout** command with any number of ephone hunt groups, but any ephone-dn to which the **auto logout** command applies must belong to only one ephone. Automatic Agent Status Not-Ready is not supported on shared lines.

Examples

This section provides the following examples:

- Cisco CME 3.3 and Earlier Versions
- Cisco Unified CME 4.0 and Later Versions

Cisco CME 3.3 and Earlier Versions

In the following example, ephone hunt group 1 is configured to permit automatic logout. If hunt group calls that are presented to 1001 and 1002 are unanswered (that is, if they ring longer than 40 seconds each), ephone 1 and ephone 2 are automatically put into DND mode. All unanswered calls are sent to voice mail (5000).

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1001
Router(config)# ephone-dn 2
```

```
Router(config-ephone-dn)# number 1002
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002
Router(config-ephone-hunt)# final 5000
Router(config-ephone-hunt)# timeout 40
Router(config-ephone-hunt)# auto logout
Router(config)# ephone 1
Router(config-ephone)# button 1:1
Router(config)# ephone 2
Router(config-ephone)# button 1:2
```

Cisco Unified CME 4.0 and Later Versions

In the following example, Automatic Agent Status Not-Ready is limited to dynamic hunt group members who do not answer two consecutive ephone hunt group calls. Ephone-dn 33, extension 1003, has dynamically joined ephone-hunt group 1. Ephone 3 will be put into DND mode if extension 1003 does not answer two consecutive hunt group calls. Ephones 1 and 2 will not be put into DND if they do not answer hunt-group calls, because the **auto logout** command applies only to dynamic hunt-group agents.

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group logout DND
Router(config)# ephone-dn 11
Router(config-ephone-dn)# number 1001
Router(config)# ephone-dn 22
Router(config-ephone-dn)# number 1002
Router(config)# ephone-dn 33
Router(config-ephone-dn)# number 1003
Router(config-ephone-dn)# ephone-hunt login
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, *
Router(config-ephone-hunt)# final 5000
Router(config-ephone-hunt)# auto logout 2 dynamic
Router(config)# ephone 1
Router(config-ephone)# button 1:11
Router(config)# ephone 2
Router(config-ephone)# button 1:22
Router(config)# ephone 3
Router(config-ephone)# button 1:33
```

In the following example, Automatic Agent Status Not-Ready cannot be used because all of the ephone-dns are shared.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1001
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 1002
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002
Router(config-ephone-hunt)# final 6000
Router(config)# ephone 1
Router(config-ephone)# button 1o1,2
Router(config)# ephone 2
Router(config-ephone)# button 1o1,2
```

Related Commands

Command	Description
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.
timeout	Defines the number of seconds after which a call that is not answered is redirected to the next number in a Cisco Unified CME ephone-hunt-group list.

auto logout (voice hunt-group)

To enable the automatic change of a voice hunt group agent's voice register dn or ephone-dn to not-ready status after a specified number of hunt-group calls are not answered, use the **auto logout** command in voice hunt group configuration mode. To disable automatic logout, use the **no** form of this command.

auto logout [*number-of-calls*]
no auto logout [*number-of-calls*]

Syntax Description

number-of-calls	Number of unanswered hunt-group calls to a voice register dn or ephone-dn before the DN is automatically changed to not-ready status. Range is 1 to 20. Default is 1.

Command Default

Automatic change of agent status to not-ready is disabled.

Command Modes

voice hunt-group configuration (config-voice-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1 15.6(3)M1	Cisco Unified CME 11.6	This command was introduced.

Usage Guidelines

This command is used with the Automatic Agent Status Not-Ready feature for voice hunt groups, which automatically puts an agent's phone in not-ready status when it exceeds a specified limit. The limit at which the Automatic Agent Status Not-Ready feature is triggered has to be specified from the range of 1 to 20. If no value is defined, the default value of 1 is applied. When an agent returns to ready status, the voice hunt group resumes sending calls to the agent's DN.

When Automatic Agent Status Not-Ready is specified for a voice hunt group and it is triggered because a DN member does not answer a specified number of voice hunt group calls, the following actions take place:

- If the **hunt-group logout HLog** command is configured, then the DNs of that hunt group switch to not-ready state when the number of successive unanswered hunt group calls specified under **auto logout** command is matched. When **hunt-group logout HLog** command is configured, phone level logout happens for SIP phones. However, SCCP phones log out at line level. An agent in not-ready status can return to ready status by pressing the HLog soft key, HLog feature access code (FAC), or Feature Button.
- If the **hunt-group logout DND** command is configured, then phone switches to DND mode and logs out the member when the number of successive unanswered hunt group calls specified under **auto logout** command is matched. For **hunt-group logout DND** command, both SIP and SCCP phones logout at phone level. An agent in not-ready status can return to ready status by pressing the DND softkey.

Cisco Unified CME 11.6 and Later Versions

In the following example, voice hunt-group 1 is configured to permit automatic logout. If hunt group calls that are presented to 1001, 1002, 1003, and 1004 are unanswered (that is, if they ring longer than 40 seconds each), voice register pool 1, voice register pool 2, ephone 1, and ephone 2 are automatically logged out. All unanswered calls are sent to DN 5000.

```
Router(config)# voice register dn 1
Router(config-register-dn)# number 1001
Router(config)# voice register dn 2
Router(config-register-dn)# number 1002

Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1003
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 1004

Router(config)# voice register pool 1
Router(config-register-pool)# number 1 dn 1

Router(config)# voice reister pool 2
Router(config-register-pool)# number 1 dn 2

Router(config)# ephone 1
Router(config-ephone)# button 1:1

Router(config)# ephone 2
Router(config-ephone)# button 1:2

Router(config)# voice hunt-group 1 peer
Router(config-voice-hunt)# pilot 1111
Router(config-voice-hunt)# list 1001, 1002, 1003, 1004
Router(config-voice-hunt)# final 5000
Router(config-voice-hunt)# timeout 40
Router(config-voice-hunt)# auto logout 4
```

If **hunt-group logout HLog** is configured under telephony-service for this example, then the DN that does not answer 4 successive calls switches to not-ready state.

Related Commands

Command	Description
auto logout	Enables automatic change of an ephone hunt group agent's ephone-dn to not-ready status after a specified number of hunt-group calls are not answered.

auto-answer

To enable the intercom auto-answer feature on a SIP phone extension, use the **auto-answer** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

auto-answer
no auto-answer

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command creates an IP phone line connection that resembles a private line, automatic ring-down (PLAR). The auto-answer causes an extension (directory number) to operate in auto-dial fashion for outbound calls and auto answer-with-mute for inbound calls. If an extension is configured for intercom operation, it can be associated with one Cisco IP phone only.

Any caller can dial an intercom extension, and a call to an intercom extension that is originated by a nonintercom caller triggers an automatic answer exactly like a legitimate intercom call. To prevent nonintercom originators from manually dialing an intercom destination, you can use alphabetic characters when you assign numbers to intercom extensions by using the **number** command. These characters cannot be dialed from a normal phone but can be dialed by preprogrammed intercom extensions when calls are made by the router.

Use the **reset** command to reset an individual SIP phone after you make changes to an extension for a SIP phone in Cisco CME.

Examples

The following example shows how to set the auto-answer feature on SIP phone directory number 1:

```
Router(config)# voice register dn 1
Router(config-register-dn) number A5001
Router(config-register-dn) auto-answer
```

Related Commands

Command	Description
number (voice register dn)	Associates a telephone or extension number with a directory number.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
reset (voice register pool)	Performs a complete reboot of a single SIP phone associated with a Cisco CME router.

auto-line

To enable automatic line selection on an IP phone in a Cisco CallManager Express (Cisco CME) system, use the **auto-line** command in ephone configuration mode. To disable automatic line selection, use the **no** form of this command.

auto-line [{*button-number* [**answer-incoming**] | **incoming**}]
no auto-line

Syntax Description

button-number	(Optional) Selects the line associated with the specified button when the handset is lifted.
answer-incoming	(Optional) Enables automatic line selection for incoming calls on the line associated with the *button-number* argument.
incoming	(Optional) Enables automatic line selection for incoming calls only.

Command Default

Automatic line selection is enabled.

Command Modes

Ephone configuration (config ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The *button-number* argument was added.
12.4(4)XC	Cisco Unified CME 4.0	The **answer-incoming** keyword was added.
12.4(9)T	Cisco Unified CME 4.0	The **answer-incoming** keyword was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

Use the **auto-line** command with no keyword or argument enables automatic line selection on the specified ephone. Picking up a handset answers the first ringing line or, if no line is ringing, selects the first idle line. This is also the default behavior if this command is not used.

Use the **auto-line incoming** command enables automatic line selection for incoming calls only. Picking up the handset answers the first ringing line and, if no line is ringing, does not select an idle line for an outgoing call. Pressing a line button selects a line for an outgoing call.

Use the **auto-line** command with the *button-number* argument specifies the line that will automatically be selected when the handset is picked up to make an outgoing call. If a button number is specified and the line associated with that button is unavailable (because it is a shared line in use on another phone), no dial tone is heard when the handset is lifted. You must press an available line button to make an outgoing call. Incoming calls must be answered by pressing the Answer soft key or pressing the ringing line button.

Use the **answer-incoming** keyword with the *button-number* argument enables automatic line selection for incoming calls on the specified button. Picking up the handset answers the incoming call on the line button associated with the *button-number* argument.

Use the **no auto-line** command disables automatic line selection on the ephone that is being configured. Pressing the Answer soft key answers the first ringing line, and pressing a line button selects a line for an outgoing call. Picking up the handset does not answer calls or provide dial tone.

Examples

The following example shows how to disable automatic line selection. The phone user must use the Answer soft key or press a line button to answer calls, or the phone user must press a line button to initiate outgoing calls.

```
Router(config)# ephone 23
Router(config-ephone)# no auto-line
```

The following example shows how to enable automatic line selection for incoming calls only. The phone user picks up the handset to answer the first ringing line. To make outgoing calls, the phone user must press a line button.

```
Router(config)# ephone 24
Router(config-ephone)# auto-line incoming
```

The following example shows how to enable the automatic selection of line button 3 for outgoing calls when the handset is lifted. There is no automatic answering of incoming calls; the user presses the Answer soft key or presses a line button to answer a call.

```
Router(config)# ephone 26
Router(config-ephone)# auto-line 3
```

The following example shows how to enable the automatic selection of line button 3 when the handset is lifted to answer incoming calls or to make outgoing calls.

```
Router(config)# ephone 26
Router(config-ephone)# auto-line 3 answer-incoming
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.

auto-network-detect

To enable phones to automatically detect whether they are inside the corporate network or not, use the **auto-network-detect** command in vpn-profile configuration mode.

auto-network-detect [{*enabledisable*}]

Syntax Description

enable	Enables auto-network detection option for a vpn-profile.
disable	Disables automatic network detection option for a vpn-profile.

Command Default

Auto-network-detect is disabled.

Command Modes

Vpn-profile configuration (conf-vpn-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to configure automatic network detection parameter in phones. The auto-network-detect command enables phones to automatically detect whether they are inside the corporate network or not. When the auto-network detection is enabled, the phone dectects the corporate network and does not require a VPN connection to start functioning. Automatic network detection is disabled by default.

Examples

The following example shows auto-network-detect enabled for vpn-profile 1:

```
Router# show run
!
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  keepalive 50
  auto-network-detect enable
  host-id-check disable
 vpn-profile 2
  mtu 1300
  password-persistent enable
  host-id-check enable
 vpn-profile 4
  fail-connect-time 50
 sip
!
!
```

Related Commands

Command	Description
vpn-profile	Defines a VPN-profile.

auto-register

To enable automatic registration of SIP phones with the Cisco Unified CME system, use the **auto-register** command in voice register global configuration mode. This command is the parent CLI, and is used to enter into the auto registration configuration mode. To disable automatic registration of SIP phones, use the **no** form of this command.

auto-register
no auto-register

Syntax Description

auto-register	Enters auto registration mode for SIP phones registering on Unified CME.

Command Default

By default, this command is enabled.

Command Modes

voice register global configuration (config-voice-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

This command is enabled by default and allows automatic registration of SIP phones on Unified CME, provided the administrator configures the password and DN range using the relevant sub-mode CLI options. It is mandatory that the password is configured before assigning the DN range. The command **service-enable** is enabled by default when **service-enable** is enabled.

The no form of this command disables the auto registration of phones, and removes the password and DN range configurations. To disable auto registration temporarily without losing the configurations such as password and DN range, use the sub-mode CLI option, *no* **service-enable**.

Examples

The following example shows how to temporarily disable auto registration using the no form of the sub-mode option, service-enable:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)# ?

VOICE auto register configuration commands:
auto-assign Define DN range for auto assignment
default Set a command to its defaults
exit Exit from voice register group configuration mode
no Negate a command or set its defaults
password Default password for auto-register phones
service-enable Enable SIP phone Auto-Registration
template Default template for auto-register phones
```

Related Commands

Command	Description
service-enable (auto-register)	Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.
password (auto-register)	Configures the mandatory password that administrator sets for auto registration of SIP phones on Unified CME.
auto-assign (auto-register)	Configures the mandatory range of directory numbers for phones auto registering on Unified CME.
template (auto-register)	Creates a basic configuration template that supports all the configurations available on the voice register template.
auto-reg-ephone	Enables automatic registration of ephones with the Cisco Unified CME system.

auto-reg-ephone

To enable automatic registration of ephones with the Cisco Unified CME system, use the **auto-reg-ephone** command in telephony-service configuration mode. To disable automatic registration, use the **no** form of this command.

auto-reg-ephone
no auto-reg-ephone

Syntax Description

This command has no keywords or arguments.

Command Default

Automatic registration is enabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

This command is enabled by default and allows automatic registration, in which Cisco Unified CME allocates an ephone slot to any ephone that connects to it, regardless of whether the ephone appears in the configuration or not.

The **no** form of this command blocks the automatic registration of ephones whose MAC addresses are not explicitly listed in the configuration. When automatic registration is blocked, Cisco Unified CME records the MAC addresses of phones that attempt to register but cannot because they are blocked.

Use the **show ephone attempted-registrations** command to view the list of phones that have attempted to register but have been blocked. Use the **clear telephony-service ephone-attempted-registrations** command to clear the list of phones that have attempted to register but have been blocked.

Examples

The following example disables automatic registration of ephones that are not listed in the configuration:

```
Router(config)# telephony-service
Router(config-telephony)# no auto-reg-ephone
```

Related Commands

Command	Description
clear telephony-service ephone-attempted-registrations	Empties the log of ephones that unsuccessfully attempt to register with Cisco Unified CME.
show ephone attempted-registrations	Displays the log of ephones that unsuccessfully attempt to register with Cisco Unified CME.

Cisco Unified CME Commands: B

b2bua

To configure a dial peer associated with an individual Session Initiation Protocol (SIP) phone in Cisco Unified CME or a group of phones in a Cisco Unified SIP Survivable Remote Site Telephony (SRST) environment to point to Cisco Unity Express, use the **b2bua** command in dial-peer configuration mode. To disable B2BUA call flow on the dial peer, use the **no** form of this command.

b2bua
no b2bua

Syntax Description

This command has no arguments or keywords.

Command Default

B2BUA callflow is disabled.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

Use the **b2bua** command to set the Cisco Unified CME source address as the 302 redirect contact address for all calls forwarded to Cisco Unity Express.

Note Use the **b2bua** command to configure Cisco SIP SRST 3.4 only after using the **allow-connections** command to enable B2BUA call flow on the SRST gateway.

Examples

The following example shows b2bua included in the configuration for voice dial peer 1:

```
dial-peer voice 1 voip
 destination-pattern 4...
 session target ipv4:10.5.49.80
 session protocol sipv2
 dtmf-relay sip-notify
 b2bua
```

Related Commands

Command	Description
allow-connections	Enables calls between SIP endpoints in a VoIP network.
dial-peer voice	Defines a dial peer and enters dial-peer configuration mode.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
show dial-peer voice	Displays information for dial peers.

Command	Description
source-address (voice register global)	Identifies the IP address and port through which SIP phones communicate with a Cisco Unified CME router.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.

background save interval

To set the interval of the background save process, use the background save interval command in telephony-service configuration mode.

background save interval interval minutes

Syntax Description

interval minutes	Interval value in minutes. Range:1 to 1440. Must be in increments of 10.

Command Default

The default interval is 10 minutes.

Command Modes

Telephony-service configuration mode

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.

Usage Guidelines

Use this command to define the background saving interval. The configured interval value should be in increments of 10 minutes. If 0 is configured as the interval, no backup will be created. The default interval is 10 minutes.

Examples

The following example shows background save interval command configured under telephony-service configuration:

```
(config-telephony)#background
(config-telephony)#background save
(config-telephony)#background save interval
(config-telephony)#background save interval 20
(config-telephony)#background save interval 20 minutes
```

bandwidth video tias-modifier

To set the maximum video bandwidth bytes per second (bps) for SIP IP phones, use the bandwidth video tias-modifier command in voice register global configuration mode. To reset the maximum video bandwidth for SIP phones, use the **no** form of this command.

bandwidth video tias-modifier bandwidth value [**negotiate end-to-end**]
no bandwidth video tias-modifier

Syntax Description

bandwidth value	Bandwidth value in bps. Range:1 to 99999999.
negotiate end-to-end	Negotiate the minimum SIP-line video bandwidth in SDP end-to-end.

Command Default

No default bandwidth is set.

Command Modes

Voice register global

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.

Usage Guidelines

Use this command to set the maximum video bandwidth for SIP IP phones. Video calls require much higher bandwidth usage than audio only calls.When there is a limitation of resources, video call bandwidth control becomes very crucial for the system. Using the bandwidth video tias-modifier command, video calls on Cisco Unified IP Phones 9951 and 9971 can use up to 1Mbps for VGA quality video.

Examples

The following example shows bandwidth video tias modifier command configured under voice register global:

```
Router#show run
!
!
!
voice service voip
 allow-connections sip to sip
!
!
voice register global
 mode cme
 source-address 10.100.109.10 port 5060
 bandwidth video tias-modifier 256 negotiate end-to-end
 max-dn 200
 max-pool 42
 create profile sync 0004625832149157
!
voice register pool  1
 id mac 1111.1111.1111
 camera
```

Related Commands

Command	Description
video	Enables video capability on Cisco Unified SIP IP Phones 9951 and 9971.

bind

To bind the source address for signaling and media packets to the IPv4 or IPv6 address of a specific interface, use the **bind** command in SIP configuration mode. To disable binding, use the **no** form of this command.

bind {**control** | **media** | **all**} **source-interface** *interface-id* [{**ipv4-address** *ipv4-address* | **ipv6-address** *ipv6-address*}]
no bind

Syntax Description

control	Binds Session Initiation Protocol (SIP) signaling packets.
media	Binds only media packets.
all	Binds SIP signaling and media packets. The source address (the address that shows where the SIP request came from) of the signaling and media packets is set to the IPv4 or IPv6 address of the specified interface.
source-interface	Specifies an interface as the source address of SIP packets.
interface-id	Specifies one of the following interfaces: • **Async** : ATM interface • **BVI** : Bridge-Group Virtual Interface • **CTunnel** : CTunnel interface • **Dialer** : Dialer interface • **Ethernet** : IEEE 802.3 • **FastEthernet** : Fast Ethernet • **Lex** : Lex interface • **Loopback** : Loopback interface • **Multilink** : Multilink-group interface • **Null** : Null interface • **Serial** : Serial interface (Frame Relay) • **Tunnel** : Tunnel interface • **Vif** : PGM Multicast Host interface • **Virtual-Template** : Virtual template interface • **Virtual-TokenRing** : Virtual token ring
ipv4-address *ipv4-address*	(Optional) Configures the IPv4 address. Several IPv4 addresses can be configured under one interface.
ipv6-address *ipv6-address*	(Optional) Configures the IPv6 address under an IPv4 interface. Several IPv6 addresses can be configured under one IPv4 interface.

Command Default

Binding is disabled.

Command Modes

SIP configuration (conf-serv-sip)

Voice class tenant.

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.2(2)XB2	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. This command does not support the Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 in this release.
12.3(4)T	The **media** keyword was added.
12.4(22)T	Support for IPv6 was added.
Cisco IOS XE Release 2.5	This command was integrated into Cisco IOS XE Release 2.5
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

Usage Guidelines

Async, Ethernet, FastEthernet, Loopback, and Serial (including Frame Relay) are interfaces within the SIP application.

If the **bind** command is not enabled, the IPv4 layer still provides the best local address.

Examples

The following example sets up binding on a SIP network:

```
Router(config)# voice serv voip
Router(config-voi-serv)# sip
Router(config-serv-sip)# bind control source-interface FastEthernet 0
```

Related Commands

Command	Description
sip	Enters SIP configuration mode from voice service VoIP configuration mode.

blf-speed-dial

To enable Busy Lamp Field (BLF) monitoring for a speed-dial number on a phone registered to Cisco Unified CME, use the **blf-speed-dial** command in ephone or voice register pool configuration mode. To disable BLF monitoring for speed-dial, use the **no** form of this command.

blf-speed-dial *tag* *number* **label** *string* [**device**]
no blf-speed-dial *tag*

Syntax Description

tag	Number that identifies the speed-dial index. Range is 1 to 75 (Skinny Client Control Protocol, SCCP); 1 to 113 (Session Initiation Protocol, SIP).
number	Telephone number to speed dial.
label *string*	Alphanumeric label that identifies the speed-dial button. The string can contain up to 30 characters.
device	(Optional) Enables phone-based monitoring.

Command Default

BLF monitoring is disabled.

Command Modes

Ephone configuration (config-ephone)
Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(22)YB	Cisco Unified CME 7.1	This command was modified to add the **device** keyword.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.
15.2(4)M	Cisco Unified CME 9.1	This command was modified to increase the BLF speed-dial index for Cisco Unified SIP phones to 113.

Usage Guidelines

This command enables a phone to monitor the status of a line associated with a speed-dial button. The **device** keyword enables BLF monitoring of the phone for which the watched directory number is the primary line. This allows watchers to monitor whether a user is on the phone, not just on an individual line on the phone.

The directory number associated with the speed-dial number must have presence enabled with the **allow watch** command. For device-level monitoring, all directory numbers associated with the monitored phone require the **allow watch** command. If any of the directory numbers is missing this configuration, the device status reported to the watcher could be inconsistent.

After using the **blf-speed-dial** command for Cisco Unified SIP IP phones, you must generate a new configuration profile using the **create profile** command and then restart the phones with the **restart** command.

For information on the BLF status indicators that display on specific types of phones in Cisco Unified CME, see the Cisco Unified IP Phone documentation for your phone model.

Examples

The following example shows BLF speed-dial monitoring enabled on phone 1 for individual directory numbers. The line status of extensions 51212 and 51214 displays on phone 1 show that presence is enabled for those directory numbers.

```
Router(config)# ephone 1
Router(config-ephone)# blf-speed-dial 1 51212 label sales
Router(config-ephone)# blf-speed-dial 2 51214 label payroll

Router(config)# voice register pool 1
Router(config-register-pool)# blf-speed-dial 1 51212 label sales
Router(config-register-pool)# blf-speed-dial 2 51214 label payroll
```

The following example shows phone-based BLF speed-dial monitoring enabled on phone 2. The line status of all extensions on the phone for which 51212 is the primary number display shows that presence is enabled for those directory numbers.

```
Router(config)# ephone 2
Router(config-ephone)# blf-speed-dial 1 51212 label sales device

Router(config)# voice register pool 2
Router(config-register-pool)# blf-speed-dial 1 51212 label sales device
```

The following example shows BLF speed-dial monitoring enabled on key 13 of phone 3:

```
Router(config)# voice register pool 3
Router(config-register-pool)# blf-speed-dial 13 51212 label sales device
```

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
create profile	Generates the configuration profile files required for Cisco Unified SIP IP phones.
presence	Enables presence service and enters presence configuration mode.
presence call-list	Enables BLF monitoring for call lists and directories on phones registered to a Cisco Unified CME router.
restart (voice register)	Performs a fast restart of one or all Cisco Unified SIP IP phones associated with a Cisco Unified CME router.

bnea

To specify the audio file used for the busy station not equipped for preemption announcement, use the **bnea** command in voice MLPP configuration mode. To disable use of this audio file, use the **no** form of this command.

bnea *audio-url*
no bnea

Syntax Description

audio-url	Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.

Command Default

No announcement is played.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Release	Cisco Products	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or. au format) for the announcement that is played to the caller when the dialed number is not preemptable.

The **mlpp indication** command must be enabled (default) for a phone to play preemption announcements.

This command is not supported by Cisco IOS help. If you type **?**, Cisco IOS help does not display a list of valid entries.

Examples

The following example shows the busy station not equipped for preemption announcement is set to the file named bnea.au located in flash:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# bnea flash:bnea.au
```

Related Commands

Command	Description
access-digit	Defines the access digit that phone users dial to request a precedence call.
bpa	Specifies the audio file used for the blocked precedence announcement.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.

bpa

To specify the audio file used for the blocked precedence announcement, use the **bpa** command in voice MLPP configuration mode. To disable use of this audio file, use the **no** form of this command.

bpa *audio-url*
no **bpa**

Syntax Description

audio-url	Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.

Command Default

No announcement is played.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Release	Cisco Products	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that is played to the caller in the following situations:

- Destination party for the precedence call is off hook.
- Destination party is busy with a precedence call of an equal or higher precedence and the destination party does not have Call Waiting or Call Forward configured, and does not have an attendant-console service configured.

The **mlpp indication** command must be enabled (default) for a phone to play precedence announcements.

This command is not supported by Cisco IOS help. If you type **?** , Cisco IOS help does not display a list of valid entries.

Examples

The following example shows the blocked precedence announcement is set to the file named bpa.au located in flash:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# bpa flash:bpa.au
```

Related Commands

Command	Description
attendant-console	Specifies the phone number of the MLPP attendant-console service.
bnea	Specifies the audio file used for the busy station not equipped for preemption announcement.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.

Command	Description
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.

bulk

To set bulk registration for E.164 numbers that will register with SIP proxy server, use the **bulk** command in voice register global configuration mode. To disable bulk registration, use the **no** form of this command.

bulk *number-pattern*
no bulk

Syntax Description

number-pattern	A sequence of digits including wild card character.

Command Default

Bulk registration is disabled.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command allows you to configure bulk registration for registering a block of phone numbers with an external registrar so that calls can be routed to Cisco CME from the SIP network.

Numbers that match the number pattern defined by using the **bulk** command register with the external registrar. The block of numbers that is registered can include any phone that is attached to Cisco CME using SIP or SCCP, or any analog phone that is directly attached to a Cisco router FXS port.

A number can contain one or more periods (.) as wildcard characters that will match any dialed number in that position. For example, 51.. rings when 5100 is dialed, when 5101 is dialed, and so forth.

The external registrar is configured by using the **registrar server** command under the SIP user-agent configuration mode.

Examples

The following example shows how to specify that numbers matching 1235 and any other dialed number in the next four positions, be routed to the Cisco CME from the SIP network.

```
Router(config)# voice register global
Router(conf-register-global)# mode cme
Router(conf-register-global)# bulk 1235...
```

Related Commands

Command	Description
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.
no reg (voice register dn)	Specifies that a directory number in a SIP Cisco CallManager Express (Cisco CME) system not register with an external proxy server
no reg (voice hunt-group)	Specifies that a pilot number for a voice hunt group not register with an external proxy server

Command	Description
registrar	Enables SIP registrar functionality.

bulk-speed-dial prefix

To set the prefix code that phone users dial to access speed-dial numbers from a global bulk speed-dial list, use the **bulk-speed-dial prefix** command in telephony-service configuration mode. To return the prefix code to the default, use the **no** form of this command.

bulk-speed-dial prefix *prefix-code*
no bulk-speed-dial-prefix

Syntax Description

prefix-code	One to four-character access code for speed dial. Default is #.

Command Default

The default prefix code (number sign [#]) is used.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

This command changes the prefix code that a phone user must dial to access speed-dial numbers from a speed-dial list that is enabled using the **bulk-speed-dial list** command in telephony-service configuration mode. The default prefix is # (number sign).

If a bulk speed-dial list is enabled using this command in telephony-service configuration mode and is also enable using this command in ephone configuration mode, the list enabled in ephone configuration mode takes precedence over the list at the global level for a given prefix. However, if the prefix used at the global level is different than the prefix used at the phone level, the lists are treated as separate lists - each list being associated with a different prefix, and at the phone level, you can access both lists.

Use the **show telephony-service bulk-speed-dial** to display information about bulk speed-dial lists that are configured in Cisco Unified CME.

Examples

The following example changes the default bulk speed-dial prefix to #7 and enables global bulk speed-dial list number 6 for all phones. It also enables a personal bulk speed-dial list for ephone 2. In this example, ephone 2 can access all of the numbers in both lists because each list is assigned a different prefix (# and #7).

```
telephony-service
 bulk-speed-dial list 6 flash:sd_dept_01_1_87.txt
 bulk-speed-dial prefix #7
ephone-dn 3
 number 2555
ephone-dn 4
 number 2557
ephone 2
 button 1:3 2:4
 bulk-speed-dial list 7 flash:lmi_sd_list_08_24_95.csv
```

Related Commands

Command	Description
bulk-speed-dial list	Enables a bulk speed-dial list.
show telephony-service bulk-speed-dial	Displays information about bulk speed-dial lists that are configured in Cisco Unified CME.

busy-trigger-per-button

To set the maximum number of calls allowed on an octo-line directory number before activating Call Forward Busy or a busy tone, use the **busy-trigger-per-button** command in ephone or ephone-template configuration mode. To reset to the default, use the **no** form of this command.

busy-trigger-per-button *number-of-calls*
no busy-trigger-per-button

Syntax Description

number-of-calls	Maximum number of calls. Range: 1 to 8. Default: 0 (disabled).

Command Default

Disabled (busy trigger is 0).

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command limits the calls to an octo-line on the specified phone by triggering Call Forward Busy or a busy tone. After the number of active calls, incoming and outgoing, on an octo-line directory number reaches the limit set with this command, the next incoming call to the directory number is forwarded to the Call Forward Busy destination. If Call Forward Busy is not configured, Cisco Unified CME rejects the call and plays a busy tone.

This command applies to each octo-line directory number on the phone.

If a directory number is shared among different phones, the busy trigger is initiated after the number of existing calls exceeds the limit set on any of the phones that share the directory number.

This command must be set to a value that is less than or equal to the value set with the **max-calls-per-button** command.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode for the same ephone, the value that you set in ephone configuration mode has priority.

Examples

The following example shows that after an octo-line on ephone 1 receives four calls, the fifth incoming call triggers Call Forward Busy or a busy tone.

```
Router(config)#
ephone 1
Router(config-ephone)# busy-trigger-per-button 4
```

Related Commands

Command	Description
call-forward busy	Enables call forwarding so that incoming calls to a busy extension are forwarded to another extension.
ephone-dn	Configures a directory number for SCCP phones.
max-calls-per-button	Sets the maximum number of calls allowed on an octo-line directory number on an SCCP phone.

busy-trigger-per-button (voice register pool)

To set the maximum number of calls allowed on a SIP directory number before activating Call Forward Busy or a busy tone, use the **busy-trigger-per-button** command in voice register pool configuration mode. To reset to the default, use the **no** form of this command.

busy-trigger-per-button *number*
no busy-trigger-per-button

Syntax Description

number	Maximum number of calls. Range: 1 to 50.

Command Default

no busy-trigger-per-button

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command limits the number of calls to each directory number on the specified phone by triggering Call Forward Busy or a busy tone. After the number of active calls, both incoming and outgoing, reaches the number of calls set with this command, Cisco Unified CME forwards the next incoming call to the Call Forward Busy destination. Cisco Unified CME rejects the call and plays a busy tone if Call Forward Busy is not configured.

If a directory number is shared among different phones, the busy trigger is initiated after the number of existing calls exceeds the limit set on all of the phones that share the directory number.

This command must be set to a value that is less than or equal to the value set with the **max-calls-per-button** command.

Examples

The following example shows that after a shared-line on phone 1 receives four calls, the fifth incoming call triggers Call Forward Busy or a busy tone.

```
Router(config)#
voice register pool 1
Router(config-register-pool)# busy-trigger-per-button 4
```

Related Commands

Command	Description
huntstop (voice register dn)	Disables call hunting behavior for a directory number on a SIP phone.
max-calls-per-button	Sets the maximum number of calls allowed on an octo-line directory number on an SCCP phone.
shared-line	Creates a directory number to be shared by multiple SIP phones.

button

To associate ephone-dns with individual buttons on a Cisco Unified IP phone and to specify line type or ring behavior, use the **button** command in ephone configuration mode. To remove an ephone-dn association from a button, use the **no** form of this command.

button *button-number {separator} dn-tag [,dn-tag...] [button-number{x}overlay-button-number] [button-number...]*
no button *button-number {separator} dn-tag [,dn-tag...] [button-number{x}overlay-button-number] [button-number...]*

Syntax Description

button-number	Number of a line button on a Cisco Unified IP phone that is to be associated with an extension (ephone-dn). The maximum number of button–ephone-dn pairs is determined by the phone type. **Note** The Cisco Unified IP Phone 7910G has only one physical line button, but you can assign it two button–ephone-dn pairs.
separator	Single character that denotes the characteristics to be associated with this phone button. Valid entries are as follows: • **:** (colon)—Normal ring. For incoming calls on this extension, the phone produces audible ringing, a flashing icon in the phone display, and a flashing red light on the handset. On the Cisco IP Phone 7914 Expansion Module, a flashing yellow light also accompanies incoming calls. • **b**—Beep but no ring. Audible ring is suppressed for incoming calls, but call-waiting beeps are allowed. Visible cues are the same as those described for a normal ring. • **c**—Call waiting. Provides call waiting for secondary calls to an overlaid ephone-dn. See also the **o** keyword. • **f**—Feature ring. Differentiates incoming calls on a special line from incoming calls on other lines on the phone. The feature-ring cadence is a triple pulse, as opposed to a single pulse for normal internal calls and a double pulse for normal external calls.
	• **m**—Monitor mode for a shared line. Visible line status indicates whether the line is in-use or not. Monitored lines cannot be used on this phone for incoming or outgoing calls. • **o**—Overlay line. Multiple ephone-dns share a single button, up to a maximum of 25 on a button. See also the **c** keyword. • **s**—Silent ring. Audible ring and call-waiting beep are suppressed for incoming calls. The only visible cue is a flashing ((< icon in the phone display. **Note** In Cisco IOS Release 12.4(4)XC and later releases, the silent ringing behavior is overridden during active night-service periods. Silent ringing does not apply during designated night-service periods when the **s** keyword is used. • **w**—Watch mode for all lines on the phone for which this directory number is the primary line. Visible line status indicates whether watched phone is idle or not.

dn-tag	Ephone-dn tag that was previously defined using the **ephone-dn** command. When used with the **c** and **o** keywords, the *dn-tag* argument can contain up to 25 individual dn-tags, separated by commas.
x	Separator that creates an overlay rollover button. When the overlay button specified in this command is occupied by an active call, a second call to one of its ephone-dns will appear on this button. This button is also known as an overlay expansion button.
overlay-button-number	Number of the overlay button that should overflow to this button.

Command Default

No buttons are defined for an ephone.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(11)YT	Cisco ITS 2.1	The **b** and **s** keywords were added.
12.2(15)ZJ	Cisco CME 3.0	The **f**, **m**, and **o** keywords were added.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)XL	Cisco CME 3.2.1	The **c** keyword was added and the ability to use the **m** keyword to monitor call-park slots was added.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The **x** keyword was added and the number of ephone-dns that can be overlaid on a single button with the **o** or **c** keyword was increased from 10 to 25. The interaction between the keyword and night service was modified; silent ringing is overridden when night service is active.
12.4(9)T	Cisco Unified CME 4.0	The modifications made to this command were integrated into Cisco IOS Release 12.4(9)T.
12.4(11)XJ3	Cisco Unified CME 4.1	The **w** keyword was added.
12.4(15)T	Cisco Unified CME 4.1	This command with the **w** keyword was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

The **button** command assigns telephone extensions to Cisco Unified IP phones by associating a button number with one or more directory numbers (ephone-dns).

Note After adding or changing a phone button configuration using this command, you must perform a quick reboot of the phone using the **restart** command.

Telephone services such as call waiting and three-party conferences require a minimum of two phone lines (ephone-dns defined with the **ephone-dn** command) to be available and configured on a Cisco IP phone.

The Cisco Unified IP Phone 7910G has only one physical line button. To support call waiting and three-party conferences on a Cisco Unified IP Phone 7910G, a second (hidden) line is required. This line cannot be selected directly using a line button. You can access the second line when you press the Conference button. You can also support multiple-call services using the **ephone-dn dual-line** configuration option.

Feature Ring (f)

A feature ring is a third type of ring cadence, in addition to the internal call and external call ring cadences. For example, an internal call in the United States rings for 2 seconds on and 4 seconds off (single-pulse ring), and an external call rings for 0.4 seconds on, 0.2 seconds off, 0.4 seconds on, and 0.2 seconds off (double-pulse ring). A feature ring is a triple-pulse ring. The purpose of associating a feature ring with a line button is to be able to identify from a distance a special line that is ringing on a multiline phone.

Monitor Mode (m)

A line button set in monitor mode on one phone displays visual line status for a line that also appears on another phone. When monitor mode is set for a button with a shared line, the line status indicates that the shared line is either idle or in use. The line and line button are available in monitor mode for visual status only. Calls cannot be made or received using a line button that has been set in monitor mode. Incoming calls on a line button that is in monitor mode do not ring and do not display caller ID or call-waiting caller ID.

Monitor mode is intended for use only in the context of shared lines so that a receptionist can visually monitor the in-use status of several users' phone extensions (for example, as a busy-lamp field). To monitor all lines on an individual phone so that a receptionist can visually monitor the in-use status of that phone, see the Watch Mode (w) section.

The line button for a monitored line can also be used as a direct-station-select for a call transfer when the monitored line is in an idle state. In this case, the receptionist who transfers a call from a normal line can press the Transfer button and then press the line button of the monitored line, causing the call to be transferred to the phone number of the monitored line.

Overlay (o)

Overlay lines are ephone-dns that share a single button on a multibutton phone. When more than one incoming call arrives on lines that are set on a single button, the line (ephone-dn) that is the leftmost in the **button** command list is the primary line and is given the highest priority. If this call is answered by another phone or if the caller hangs up, the phone selects the next line in its overlay set to present as the ringing call. The caller ID display updates to show the caller ID for the currently presented call.

Ephone-dns that are part of an overlay set can be single-line ephone-dns or dual-line ephone-dns, but the set must contain either all single-line ephone-dns or all dual-line ephone-dns, and not a mixture of the two.

The primary ephone-dn on each phone in a shared-line overlay set should be unique to the phone being configured to guarantee that there is a line available for outgoing calls, and to ensure that the phone user can obtain dial-tone even when there are no idle lines available in the rest of the shared-line overlay set. Use a unique ephone-dn in this manner to provide for a unique calling party identity on outbound calls made by the phone so that the called user can see which specific phone is calling.

The name of the first ephone-dn in the overlay set is not displayed because it is the default ephone-dn for calls to the phone, and the name or number is permanently displayed next to the phone's button. For example, if there are ten ephone-dns in an overlay set, only the last nine ephone-dns are displayed when calls are made to them.

Overlay Ephone-dns with Call Waiting (c)

The configuration for the overlaid ephone-dns with call waiting (keyword **c**) and without call waiting (keyword **o**) is the same.

Ephone-dns accept call interruptions, such as call waiting, by default. For call waiting to work, the default must be active. To ensure thatthe default is active, remove the **no call-waiting beep accept** command from the configurations of ephone-dns for which you want to use call waiting.

In Cisco Unified CME 4.0(3), the Cisco Unified IP Phone 7931G cannot support overlays that contain ephone-dn configured for dual-line mode.

Note In general, all the ephone-dns within an overlay must be of the same type (dual-line or single line mode).

Silent Ring (s)

You can configure silent ring on any type of phone. However, you typically set silent ring only on buttons of a phone with multiple lines, such as a Cisco Unified IP Phone 7940, Cisco Unified IP Phones 7960 and 7960G, or a Cisco Unified IP Phone 7914 Expansion Module. The only visible cue is a flashing ((< icon in the phone display.

If you configure a button to have a silent ring, you will not hear a call-waiting beep or call-waiting ring regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting beep or call-waiting ring.

In Cisco IOS Release 12.4(4)XC and later releases, the silent ringing behavior is overridden during active night-service periods. Silent ringing does not apply during designated night-service periods when the **s** keyword is used.

Watch Mode (w)

A line button that is configured for watch mode on one phone provides visual line status for all lines on another phone (watched phone) for which the watched directory number is the primary line. Watched mode allows a phone user, such as a receptionist, to visually monitor the in-use status of an individual phone. The line and line button on the watching phone are available in watch mode for visual status only. Calls cannot be made or received using a line button that has been set in watch mode. Incoming calls on a line button that is in watch mode do not ring and do not display caller ID or call-waiting caller ID.

If any of the following conditions are true, the status of the line button in watch mode is that the watched phone is in-use:

- Watched phone is off-hook
- Watched phone is not registered
- Watched phone is in the do-not-disturb (DND) mode
- Watched directory number is not idle

If the watched directory number is a shared line and the shared line is not idle on any phone with which it is associated, then in the context of watch mode, the status of the line button indicates that the *watched phone* is in use.

For best results in terms of monitoring the status of an individual phone based on a watched directory number, the directory number to be configured for watch mode should not be a shared line. To monitor a shared line so that a receptionist can visually monitor the in-use status of several users' phone extensions, see the Monitor Mode (m) section.

If the watched directory number is associated with several phones, then the watched phone is the one on which the watched directory number is on button 1 or the one on which the watched directory number is on the button that is configured by using the **auto-line** command, with auto-line having priority.

If more than one phone meets the criteria for primary line as described above, then the watched phone is the first phone that that meets the criteria. Typically, that is the phone with the lowest ephone tag value. However, if the watched directory number is configured on button 1 of ephone 1 and the same directory number is also configured on button 3 with "auto-line 3" of ephone 24, then ephone 24 is the watched phone because the auto-line configuration has priority.

The line button for a watched phone can also be used as a direct-station-select for a call transfer when the watched phone is idle. In this case, the phone user who transfers a call from a normal line can press the Transfer button and then press the line button of the watched directory number, causing the call to be transferred to the phone number associated with the watched directory number.

Expansion Buttons for Overlay Ephone-dns (x)

This feature works to expand coverage for an overlay button that has been configured using the **o** separator in the **button** command. Overlay buttons with call waiting that use the **c** separator in the **button** command are not eligible for overlay rollover.

Examples

The following example assigns four button numbers on the phone to ephone-dn tags. Button 4 is configured for a silent ring:

```
ephone-dn 1
 number 233
ephone-dn 4
 number 234
ephone-dn 16
 number 235
ephone-dn 19
 number 236
ephone 1
 button 1:1 2:4 3:16 4s19
```

The following example shows three phones that each have three instances of extension number 1001 overlaid onto a single button, which allows three simultaneous calls to extension 1001. The first call arrives on ephone-dn 1 and rings button 1 on all three phones. The call is answered on ephone 10. A second call for 1001 hunts onto ephone-dn 2 and rings on the remaining two ephones, ephones 11 and 12, and is answered by ephone 12. A third call to 1001 hunts onto ephone-dn 3 and rings on ephone 12, where it is answered. This configuration creates a three-way shared line across three IP phones and can handle three simultaneous calls to the same telephone number. Note that if ephone 12 is busy, the third call will go to voice mail (7000). Note also that if you want to configure call waiting, you can use the same configuration, except use the **c** keyword instead of the **o** keyword. Ephone 10 uses call waiting.

```
ephone-dn 1
 number 1001
 no huntstop
!
ephone-dn 2
```

```
 number 1001
 no huntstop
 preference 1
!
ephone-dn 3
 number 1001
 preference 2
 call-forward busy 7000
!
! The next ephone configuration includes the first instance of shared line 1001.
ephone 10
 mac-address 1111.2222.3333
 button 1c1,2,3
!
! The next ephone configuration includes the second instance of shared line 1001.
ephone 11
 mac-address 1111.2222.4444
 button 1o1,2,3
!
! The next ephone configuration includes the third instance of shared line 1001.
ephone 12
 mac-address 1111.2222.555
 button 1o1,2,3
```

The following is an example of a unique ephone-dn as the primary dn in a simple shared-line overlay configuration. The no huntstop command is configured for all the ephone-dns except ephone-dn 12, the last one in the overlay set. Because the ephone-dns are dual-line dns, the huntstop-channel command is also configured to ensure that the second channel remains free for outgoing calls and for conferencing.

```
ephone-dn 1 dual-line
 number 101
 huntstop-channel
!
ephone-dn 2 dual-line
 number 102
 huntstop-channel
!
ephone-dn 10 dual-line
 number 201
 no hunsttop
 huntstop-channel
!
ephone-dn 11 dual-line
 number 201
 no hunsttop
 huntstop-channel
!
ephone-dn 12 dual-line
number 201
huntstop-channel
!
!The next ephone configuration includes (unique) ephone-dn 1 as the primary line in a
shared-line overlay
ephone 1
mac-address 1111.1111.1111
button 1o1,10,11,12
!
!The next ephone configuration includes (unique) ephone-dn 2 as the primary line in another
 shared-line overlay
ephone 2
mac-address 2222.2222.2222
button 1o2,10,11,12
```

Shared-line overlays can be constructed using the “button o” or “button c” formats, depending on whether call-waiting is desired. The following example shows an ephone configuration that enables call waiting (c) in a shared-line overlay:

```
ephone 1
 mac-address 1111.1111.1111
 button 1c1,10,11,12
!
 ephone 2
 mac-address 2222.2222.2222
 button 1c2,10,11,12
```

The following example configures a “3x3” shared-line setup for three ephones and nine shared lines (ephone-dns 20 through 28). Each ephone has a unique ephone-dn for each of its three buttons (ephone-dns 1 to 3, ephone-dns 4 to 6, and ephone-dns 7 to 9). The remaining ephone-dns are shared among the three phones. Three phones with three buttons each can take nine calls. The overflow buttons provide the ability for an incoming call to ring on the first available button on each phone.

```
ephone 1
 button 1o1,2,3,20,21,22,23,24,25,26,27,28 2x1 3x1
ephone 2
 button 1o4,5,6,20,21,22,23,24,25,26,27,28 2x1 3x1
ephone 3
 button 1o7,8,9,20,21,22,23,24,25,26,27,28 2x1 3x1
```

Related Commands

Command	Description
call-waiting beep	Allows phone buttons to accept or generate call-waiting beeps.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco Unified CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show ephone	Displays information about ephones and the corresponding Cisco Unified IP phones.
show ephone overlay	Displays the configuration and current status of registered overlay ephone-dns.

button-layout (voice register template)

To organize the order of the display of all buttons including line, speed dial, blf speed dial, feature buttons, and url buttons on a Cisco Unified SIP IP phone, use the **button-layout** command in voice register template configuration mode. To disable the feature button set and change the action of the buttons on IP phones, use the **no** form of this command.

button-layout [*button-string*] [*button-type*]
no button-layout

Syntax Description

button-string	(Optional) Specifies a comma-separated list of physical button number or ranges of button numbers.
button-type	(Optional) Specifies one of the following button types: Line, Speed-Dial, BLF-Speed-Dial, Feature, URL.

Command Default

No fixed set of line or feature buttons are defined.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use the button-layout command to assign physical button numbers or ranges of numbers with button types such as line, feature, url, speed-dial, and blf-speed-dIal. After creating a voice register template and applying the template to the voice register pool you can assign the button-layout configuration to a Cisco Unified IP Phone.

Note The first button needs to be the line button so that the phone can complete provisioning.

Examples

The following example shows button-layout configured on voice register template 2 and voice register template 5.

```
 Router# show voice register template all
!
voice register dn  65
 number 3065
 name SIP-7965
 label SIP3065
!
voice register template  5
 button-layout 1 line
 button-layout 2,5 speed-dial
 button-layout 3,6 blf-speed-dial
 button-layout 4,7,9 feature-button
 button-layout 8,11 url-button
!
```

```
voice register template  2
 button-layout 1,5 line
 button-layout 4 speed-dial
 button-layout 3,6 blf-speed-dial
 button-layout 7,9 feature-button
 button-layout 8,10-11 url-button
!
```

Related Commands

Command	Description
ephone-template (ephone)	Applies template to an ephone.
show voice register template	Displays all configuration information associated with a SIP phone template.

button-layout

To configure a fixed set of line or feature buttons in an ephone-template which can then be applied to a supported IP phone in Cisco Unified CME, use the **button-layout set** command in ephone-template configuration mode. To disable the feature buttons set and change the action of the buttons on IP phones, use the **no** form of this command.

button-layout [{*phone-type* {**1** | **2**} | **button-string** | **button-type**}]
no button-layout

Syntax Description

phone-type	Type of IP phone. The following choices are valid: • **7931**—Cisco Unified IP Phone 7931.
1	Number of fixed line or feature set containing the following buttons: • Button 24—Menu. • Button 23—Headset.
2	Number of fixed line or feature set containing the following buttons: • Button 24—Menu. • Button 23—Headset. • Button 22—Directories. • Button 21—Messages.
button-string	(Optional) Specifies a coma separated list of physical button number or ranges of button numbers.
button-type	(Optional) Specifies one of the following button types: Line, Speed-Dial, BLF-Speed-Dial, Feature, URL

Command Default

No fixed set of line or feature buttons are defined.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(6)XE	Cisco Unified CME 4.0(2)	This command was introduced.
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)T	Cisco Unified CME 4.0(3)	This command was integrated into Cisco IOS Release 12.4(11)T.
15.1(3)T	Cisco Unified CME 8.5	This command was modified. Button String and Button Type arguments were added.

Usage Guidelines

Use this command to configure either Set 1 or Set 2 in an ephone-template which can then be applied to an individual Cisco Unified IP Phone 7931G in Cisco Unified CME.

After a template has been created, you can apply it to an ephone using the **ephone-template** command in ephone configuration mode. You cannot apply more than one ephone template to an ephone.

To view your ephone-template configurations, use the **show telephony-service ephone-template** command.

In Cisco Unified CME 8.5 and later versions, the button-layout command allows you to assign physical button numbers or ranges of numbers with button types such as Line, Feature, URL, Speed-Dial, BLF-Speed-DIal. After creating an ephone-template you can apply the button-layout configuration to a Cisco Unified IP Phone.

Examples

1. The following example shows how to create ephone-template 12, containing set 2 feature buttons, and apply the template to ephone 36.

```
Router(config)# ephone-template 12
Router(config-ephone-template)# button-layout set 2
Router(config-ephone-template)# exit
Router(config)# ephone 36
Router(config-ephone)# ephone-template 12
Router(config-ephone)# exit
Router(config)# telephony-service
Router(config-telephony)# create cnf-files
```

1. The following example shows ephone-template 10, containing line button, speed-dial button, blf-speed-dial button, feature button, and url button.

```
Router# show telephony-service ephone-template
ephone-template  10
 button-layout 1 line
 button-layout 2,5 speed-dial
 button-layout 3,6 blf-speed-dial
 button-layout 4,7,9 feature
 button-layout 8,11 url
```

Related Commands

Command	Description
ephone-template (ephone)	Applies template to an ephone.
show telephony-service ephone-template	Displays ephone-template configurations.

Cisco Unified CME Commands: C

call application voice aa-hunt

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice aa-hunt** command is replaced by the **param aa-hunt** command. See the **param aa-hunt** command for more information.

To declare a Cisco Unified CME basic automatic call distribution (B-ACD) menu number and associate it with the pilot number of an ephone hunt group, use the **call application voice aa-hunt command in** global configuration mode. To remove the menu number and the ephone hunt group pilot number, use the **no** form of this command.

call application voice *application-name* **aa-hunt** *menu-number pilot-number*
no call application voice *application-name* **aa-hunt** *menu-number pilot-number*

Syntax Description

menu-number	Number that callers must dial to reach the pilot number of an ephone hunt group. The range is from 1to 10. The default is 1.
application-name	Application name given to the call queue script in the **call application voice command.**
pilot-number	Ephone hunt group pilot number.

Command Default

Cisco CME B-ACD menu number 1 is configured, but it has no pilot number**.**

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced with the **param aa-hunt** command.

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD call queue scripts. Up to three menu options are allowed per call queue script. You can use any of the allowable numbers in any order.

The **call application voice aa-hunt** command allows each of the menu options to be associated with the pilot number of an ephone hunt group. The menu options are announced by the en_bacd_options_menu.au audio file, which you can rerecord. When a caller presses a number, the call will go to the pilot number of an ephone hunt group so it can be transferred to one of the ephone hunt group's ephone-dns. It will not go to any other ephone hunt group. The order in which ephone-dns are selected depends on the ephone hunt group's search method, which is configured with the **ephone-hunt** command, and whether an ephone-dn is busy or not.

If only one menu option is configured, callers will hear a greeting and be transferred directly to the pilot number of the corresponding ephone hunt group. They do not have to enter a number.

The highest aa-hunt number will automatically be set to zero (0) for the operator. In the following example, aa-hunt8 supports the menu option of 0 and 8.

```
call application voice queue aa-hunt1 1111
```

```
call application voice queue aa-hunt3 3333
call application voice queue aa-hunt8 8888
```

If a phone user presses 0 or 8, their call be sent to pilot number 3333.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example associates three menu numbers with three pilot numbers of three ephone hunt groups. Pilot number 1111 is for ephone hunt group 1 (sales); 2222 is for ephone hunt group 2 (customer service); and 3333 is for ephone hunt group 3 (operator). If sales is selected from the AA menu, the call will be transferred to 1111 and sent to ephone hunt group 1's available ephone-dns (2001, 2002, 2003, 2004, 2005, 2006).

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004, 1005, 1006, 1007, 1008, 1009, 1010
Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 2222
Router(config-ephone-hunt)# list 2001, 2002, 2003, 2004, 2005, 2006
Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# pilot 3333
Router(config-ephone-hunt)# list 3001, 3002, 3003, 3004
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue aa-hunt1 1111
Router(config)# call application voice queue aa-hunt2 2222
Router(config)# call application voice queue aa-hunt3 3333
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice aa-pilot	Associates an ephone hunt group with the Cisco CME basic service's AA script by declaring the group's pilot number.
call application voice welcome-prompt	Assigns an audio file that is used by a Cisco CME B-ACD AA script for the welcome greeting.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.
pilot	Defines the ephone-dn that callers dial to reach a Cisco CME ephone hunt group.

call application voice aa-name

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice aa-name** command is not available in Cisco IOS software.

To associate the queue script for Cisco Unified CME basic automatic call distribution (B-ACD) with the Cisco Unified CME B-ACD auto-attendant (AA) script, use the **call application voice aa-name command in** global configuration mode. To remove the queue script and AA script association, use the **no** form of this command.

call application voice *application-name* **aa-name** *aa-script-name*
no call application voice *application-name* **aa-name** *aa-script-name*

Syntax Description

application-name	Application name given to the call queue script in the **call application voice command**.
aa-script-name	Application name given to the AA script in the **call application voice command**.

Command Default

No call queue script and AA script association is configured**.**

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced with the **param aa-name** command.

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD call queue scripts. Only one AA script can be associated with one call queue script.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example associates a call queue script with an AA script:

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue aa-name aa
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice service-name	Associates a Cisco CME B-ACD AA script with a Cisco Unified CME B-ACD call queue script.

call application voice aa-pilot

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice aa-pilot** command is replaced by the **param aa-pilot** command. See the **param aa-pilot** command for more information.

To assign a pilot number to the Cisco Unified CME basic automatic call distribution (B-ACD) service, use the **call application voice aa-pilot command in** global configuration mode. To remove the Cisco Unified CME B-ACD pilot number, use the **no** form of this command.

call application voice *application-name* **aa-pilot** *pilot-number*
no call application voice *application-name* **aa-pilot** *pilot-number*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice command**.
pilot-number	Pilot number for Cisco CME B-ACD.

Command Default

No Cisco Unified CME B-ACD pilot number is configured**.**

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param aa-pilot** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. Only one pilot number can be used for each Cisco Unified CME B-ACD service, and the voice ports handling AA must have dial peers that will send calls to the pilot number.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

Examples

The following example assigns 8005550100 as the pilot number to the Cisco Unified CME B-ACD service. Included in this example is the dial-peer configuration for the pilot number.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa aa-pilot 8005550100
Router(config)# dial-peer voice 1000 pots
Router(config)# incoming pilot number 8005550100
Router(config)# application aa
Router(config)# direct-inward-dial
Router(config)# port 1/0:23
Router(config)# forward digits-all
Router(config)# call application voice aa aa-pilot 80055501
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
dial-peer voice	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco Unified CME system.

call application voice call-retry-timer

Effective with Cisco IOS Release 12.3(14)T and later, the **call application call-retry-timer** command is replaced by the **param call-retry-timer** command. See the **param call-retry-timer** command for more information.

To assign the length of time that calls to Cisco Unified CME basic automatic call distribution (B-ACD) must wait before attempting to transfer to an ephone hunt group pilot number, use the **call application voice call-retry-timer command in** global configuration mode. To remove the retry time, use the **no** form of this command.

call application voice *application-name* **call-retry-timer** *seconds*
no call application voice *application-name* **call-retry-timer** *seconds*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice command**.
seconds	Number of seconds that a call must wait before attempting to transfer an ephone hunt pilot number or voice-mail pilot number. The range is from 5 to 30 seconds. The default is 15 seconds.

Command Default

The retry interval is 15 seconds.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param call-retry-timer** command

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. The following sequence of events might occur:

- An outside call comes into a system configured with Cisco CME B-ACD.
- A menu option is selected that attempts to transfer the call to an ephone hunt group pilot number.
- All of the ephone hunt group's ephone-dns are busy.

In that case, the call will wait in a queue for the period of time set by the **call application voice call-retry-timer** command and retry to the pilot number.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

Examples

The following example shows a configuration that allows outside calls to Cisco CME B-ACD to retry an ephone hunt group pilot number every 30 seconds. The example shows the configuration for one ephone hunt group, which is presented by Cisco CME B-ACD menu as the sales department and uses a simple configuration. If a caller selects the sales menu option (**ephone-hunt 1**) and all of the ephone-dns configured in the **list** command (1001, 1002, 1003, 1004) are busy, the call will wait 30 seconds and then retry the pilot number (1111) until either an ephone-dn becomes available or a configured amount of time has elapsed (see the **call application voice max-time-call-retry command).**

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa call-retry-timer 30
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco Unified CME system.
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice aa-hunt	Declares a Cisco Unified CME B-ACD menu number and associates it with the pilot number of an ephone hunt group.
call application voice aa-pilot	Associates an ephone hunt group with the Cisco Unified CME basic service's AA script by declaring the group's pilot number
call application voice max-time-call-retry	Assigns the maximum length of time for which calls to Cisco Unified CME B-ACD can stay in a call queue.

call application voice dial-by-extension-option

Effective with Cisco IOS Release 12.3(14)T and later, the **call application dial-by-extension-option** command is replaced by the **param dial-by-extension-option** command. See the **param dial-by-extension-option** command for more information.

To enable direct extension access and set the access number for Cisco Unified CME basic automatic call distribution (B-ACD), use the **call application voice dial-by-extension-option command in** global configuration mode. To disable direct dial extension access and remove the access number, use the **no** form of this command.

call application voice *application-name* **dial-by-extension** *number*
no call application voice *application-name* **dial-by-extension** *number*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice command**.
number	The single digit that callers press to be able to enter an extension number from the AA menu. The range is from 1 to 10. There is no default.

Command Default

Direct dial access is disabled. No access number is configured.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param dial-by-extension-option** command.

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. It enables the en_bacd_enter_dest.au audio file. The default announcement says, “Please enter the extension number you want to reach.” The **call application voice dial-by-extension-option** command also allows for the configuration of the number that callers must press before they can enter the extension number that they want to call.

Callers who select the extension access option can then dial any extension. If they dial an ephone hunt group ephone-dn or pilot number, their call will not be sent to the ephone hunt-group call queue.

Examples

The following example configures Cisco CME B-ACD to include an option that allows callers to press the number 4 so they can dial an extension number.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa dial-by-extension 4
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.

call application voice drop-through-option

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the c**all application voice drop-through-option** command has been replaced by the **param drop-through-option** command.

call application voice drop-through-prompt

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice drop-through-prompt** command has been replaced by the **param drop-through-prompt** command.

call application voice handoff-string

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice handoff-string** command has been replaced by the **param handoff-string** command.

call application voice max-extension-length

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice max-extension-length** command has been replaced by the **param max-extension-length** command.

call application voice max-time-call-retry

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice max-time-call-retry** command is replaced by the **param max-time-call-retry** command. See the **param max-time-call-retry** command for more information.

To assign the maximum length of time for which calls to Cisco Unified CME basic automatic call distribution (B-ACD) can stay in a call queue, use the **call application voice max-time-call-retry command in** global configuration mode. To remove the maximum length of time, use the **no** form of this command.

call application voice *application-name* **max-time-call-retry** *seconds*
no call application voice *application-name* **max-time-call-retry** *seconds*

Syntax Description

application-name	Application name given to the auto attendant (AA) script in the **call application voice command**.
seconds	Maximum length of time that the Cisco Unified CME B-ACD AA script can keep redialing an ephone hunt group pilot number. The range is from 0 to 3600 seconds. The default is 600 seconds.

Command Default

The default maximum length of time that calls can stay in a call queue is 600 seconds.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param max-time-call-retry** command.

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. The **call application voice max-time-retry** command allows you set a time limit for the redialing of pilot numbers under the following circumstances:

- An outside call comes into a system configured with Cisco Unified CME B-ACD.
- A menu option is selected that transfers the call to an ephone hunt-group pilot number.
- All of the ephone hunt group's ephone-dns are busy.
- The call is sent to a queue and tries the pilot number at intervals of time set by the **call application voice call-retry-timer** command.

When the time period set by the **call application voice max-call-retry** command expires, one of the following two events will occur:

- If a voice-mail pilot number has been configured in Cisco Unified CME and mail boxes for hunt group pilot numbers have been configured in a voice-mail application, calls will be transferred to voice mail.

- If voice mail has not been configured, a default message will be played that says, “We are unable to take your call at this time. Please try again at a later time. Thank you for calling.”

Examples

In the following example, the length of time for which calls can try to reach ephone hunt group 1 and ephone hunt group 2 is 90 seconds. If a caller selects the AA menu option for either hunt group and all of its ephone-dns configured in the **list** command are busy, the call will keep retrying the ephone hunt group’s pilot number until one of the ephone-dns is available or 90 seconds has elapsed. When 90 seconds elapses, the call will go to voice mail.

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004
Router(config)# ephone-hunt 2peer
Router(config-ephone-hunt)# pilot 2222
Router(config-ephone-hunt)# list 2001, 2002, 2003, 2004
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa max-call-retry-timer 90
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice call-retry-timer	Assigns the length of time that calls to Cisco Unified CME B-ACD must wait before attempting to transfer to an ephone hunt-group pilot number or to voice mail.
call application voice max-time-vm-retry	Assigns the maximum number of times that calls to Cisco Unified CME B-ACD can attempt to reach voice mail.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco Unified CME system.

call application voice max-time-vm-retry

Cisco IOS Release 12.3(14)T and later releases support Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice max-time-vm-retry** command has been replaced by the **param max-time-vm-retry** command.

call application voice number-of-hunt-grps

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice number-of-hunt-grps** command is replaced by the **param number-of-hunt-grps** command. See the **param number-of-hunt-grps** command for more information.

To declare the maximum number of ephone hunt-group menus supported by Cisco Unified CME basic automatic call distribution (B-ACD), use the **call application voice number-of-hunt-grps command in** global configuration mode. To remove the maximum number of ephone hunt-group menus supported by Cisco CME B-ACD, use the **no** form of this command.

call application voice *application-name* **number-of-hunt-grps** *number*
no call application voice *application-name* **number-of-hunt-grps** *number*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice command**.
number	Number of hunt groups used by the Cisco Unified CME B-ACD AA script and call queue script. The range is from 1 to 3. The default is 3.

Command Default

Three ephone hunt-group menus are supported by Cisco CME B-ACD**.**

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param number-of-hunt-grps** command.

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. The *number* argument declares the number of ephone hunt groups only. The menu option for direct extension access (see the **call application voice dial-by-extension-option** command) is not included.

Examples

The following example configures a Cisco Unified CME B-ACD call queue script to use three ephone hunt groups and one direct extension access number, making the *number* argument in the **call application voice number-of-hunt-grps** equal to 3. The **ephone-hunt** command is used to configure the three ephone hunt groups. The **call application voice dial-by-extension-option** command is used to enable direct extension access and set the access number to 1.

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004, 1005, 1006, 1007, 1008, 1009, 1010
Router(config)# ephone-hunt 2 peer
```

```
Router(config-ephone-hunt)# pilot 2222
Router(config-ephone-hunt)# list 2001, 2002, 2003, 2004, 2005, 2006
Router(config-ephone-hunt)# final 9000
Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# pilot 3333
Router(config-ephone-hunt)# list 3001, 3002, 3003, 3004
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa dial-by-extension 1
Router(config)# call application voice aa number-of-hunt-grps 3
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice dial-by-extension-option	Enables direct extension access and sets the access number for Cisco CME B-ACD.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.

call application voice queue-len

Effective with Cisco IOS Release 12.3(14)T and later, the **call application queue-len** command is replaced by the **param queue-len** command. See the **param queue-len** command for more information.

To set the maximum number of calls allowed for each ephone hunt group's call queue that is used by Cisco Unified CME basic automatic call distribution (B-ACD), use the **call application voice queue-len command in** global configuration mode. To remove the queue-length setting, use the **no** form of this command.

call application voice *application-name* **queue-len** *number*
no call application voice *application-name* **queue-len** *number*

Command Default

application-name	Application name given to the call queue script in the **call application voice command**.
number	Number of calls that can be waiting in each ephone hunt group's queue. The range is dependent on your hardware configuration. The range is from 1 to 30. The default is 10.

Command Default

Thirty calls are allowed in each call queue.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param queue-len** command.

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD call queue scripts. The following sequence of events might occur:

- An outside call comes into a system configured with Cisco Unified CME B-ACD.
- A menu option is selected that transfers the call to an ephone hunt-group pilot number.
- All of the ephone hunt group's ephone-dns are busy.

In that case, the call will be sent to a queue for that individual hunt group. The number of calls that each ephone hunt group can hold in its queue is configured by the **call application voice queue-len** command.

In the following configuration example, ephone hunt group 1 supports two ephone-dns; ephone hunt group 2 supports three ephone-dns; and the queue length is 10 for both ephone hunt groups:

```
ephone-hunt 1 peer
 pilot 1111
 list 1001, 1002
ephone-hunt 2 peer
 pilot 2222
 list 2001, 2002, 2003
call application voice queue flash:app-b-acd-x.x.x.x.tcl
call application voice callqueuescriptfilename queue-len 10
```

If ephone hunt group 1's ephone-dns are busy, ten more calls can be made to ephone hunt group 1. During that time, the calls in the queue would periodically retry the pilot numbers (**call application voice max-time-retry-timer** command) and receive secondary greetings (**call application voice second-greeting-time** command). If none of the calls has hung up or connected to an ephone-dn, the eleventh caller would hear the en_bacd_disconnect.au message and a busy signal. The default message is, "We are unable to take your call at this time. Please try again at a later time. Thank you for calling." Includes a four-second pause after the message.

For ephone hunt group 2, three calls can be connected to ephone-dns 2001, 2002, and 2003, and ten calls can be waiting in ephone hunt group 2's queue. If the status remains unchanged, the fourteenth caller hears the disconnect message and a busy signal. But if one of the earlier calls disconnects (either by leaving the queue or by ending a call), the fourteen call enters the queue.

The maximum number of calls allowed in the queues of ephone hunt groups must be based on the number of ports or trunks available. For example, if you had 20 foreign exchange office (FXO) ports and two ephone hunt groups, you could configure a maximum of ten calls per ephone hunt-group queue with the **call application voice queue-len 10** command. You could use the same configuration if you had a single T1 trunk, which supports 23 channels.

Examples

The following example configures a Cisco Unified CME B-ACD call queue script to allow a maximum of 12 calls to wait in each ephone hunt group's calling queue for ephone-dns to become available:

```
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue queue-len 12
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice call-retry-timer	Assigns the length of time that calls to Cisco CME B-ACD must wait before attempting to transfer to an ephone hunt-group pilot number or to voice mail.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.

call application voice queue-manager-debugs

Effective with Cisco IOS Release 12.3(14)T and later, the **call application queue-manager-debugs** command is replaced by the **param queue-manager-debugs** command. See the **param aa-hunt** command for more information.

To enable or disable the collection of call queue debug information from Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD), use the **call application voice queue-manager-debugs command in** global configuration mode. To remove the current setting, use the **no** form of this command with the keyword that was used in the previous occurrence of the **call application voice queue-manager-debugs command**.

call application voice *application-name* **queue-manager-debugs** [{**0** | **1**}]
no call application voice *application-name* **queue-manager-debugs** [{**0** | **1**}]

Syntax Description

application-name	Application name given to the call queue script in the **call application voice command**.
0	Disables debugging.
1	Enables debugging.

Command Default

The collection of call queue debug information from Cisco CME B-ACD is **disabled.**

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param queue-manager-debugs** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD call queue scripts. It enables the collection of data regarding call queue activity. It is used in conjunction with the **debug voip ivr script** command. Both commands must be enabled at the same time.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example configures a Cisco CME B-ACD call queue script to enable debugging for the collection of data for the **debug voip ivr script** command:

```
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue queue-manager-debugs 1
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
debug voip ivr script	Display debugging messages for IVR scripts.

call application voice second-greeting-time

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice second-greeting-time** command is replaced by the **param second-greeting-time** command. See the **param second-greeting-time** command for more information.

To set the delay before the second greeting is played after a caller joins a Cisco Unified CME basic automatic call distribution (B-ACD) calling queue and set the interval of time at which the second-greeting message is repeated, use the **call application voice second-greeting-time command in** global configuration mode. To remove the second-greeting time, use the **no** form of this command.

call application voice *application-name* **second-greeting-time** *seconds*
no call application voice *application-name* **second-greeting-time** *seconds*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice command**.
seconds	Amount of time that second-greeting message must wait before it can be played. The range is from 30 to 120 seconds. The default is 60 seconds.

Command Default

The second-greeting delay time is 60 seconds.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param second-greeting-time** command.

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. A second greeting is an audio message of up to 15 seconds in length. The default announcement is, “All agents are currently busy assisting other customers. Continue to hold for assistance. Someone will be with you shortly.” The second-greeting message is only presented to callers waiting in a CME B-ACD call queue.

The second-greeting time is clocked when the second-greeting message begins, not after it ends. For example, if the second greeting were 15 seconds in length and the configured second-greeting time were 70 seconds, the greeting would begin every 70 seconds, not 85 seconds as if to allow for the 15-second message.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example configures a Cisco Unified CME B-ACD AA script to allow a second-greeting message to be repeated every 50 seconds as long as a call is in a call queue.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice AAscriptfilename second-greeting-time 50
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.

call application voice service-name

Cisco IOS Release 12.3(14)T and later releases support Cisco CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice service-name** command has been replaced by the **param service-name** command.

call application voice voice-mail

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice voice-mail** command is replaced by the **param voice-mail** command. See the **param voice-mail** command for more information.

To assign a pilot number for the Cisco Unified CME basic automatic call distribution (B-ACD) service's voice mail, use the **call application voice voice-mail command in** global configuration mode. To remove the voice-mail pilot number, use the **no** form of the command.

call application voice *application-name* **voice-mail** *number*
no call application voice *application-name* **voice-mail** *number*

Syntax Description

application-name	Application name given to the auto attendant (AA) script in the **call application voice command**.
number	Pilot number of the voice mail to which calls to Cisco CME B-ACD will be transferred.

Command Default

No voice-mail pilot number is configured for Cisco Unified CME B-ACD**.**

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param voice-mail** command.

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco Unified CME B-ACD AA scripts. Only one pilot number is allowed per Cisco CME B-ACD service. Calls to the service will be sent to this voice mail number.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example configures a Cisco Unified CME B-ACD voice-mail pilot number as 5000.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa voice-mail 5000
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.

call application voice welcome-prompt

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice welcome-prompt** command is replaced by the **param welcome-prompt** command. See the **param welcome-prompt** command for more information.

To assign an audio file that is used by the Cisco Unified CME basic automatic call distribution (B-ACD) auto-attendant (AA) script for the welcome greeting, use the **call application welcome-prompt command in** global configuration mode. To remove the audio file assignment, use the **no** form of this command.

call application voice *application-name* **welcome-prompt _** *audio-filename*
no call application voice *application-name* **welcome-prompt _** *audio-filename*

Syntax Description

application-name	Application name given to the AA script in the **call application voice command**.
_audio-filename	Filename of the welcome greeting to be played when callers first reach the Cisco Unified CME B-ACD, preceded by the underscore (_) character. The filename must not have a language code prefix, such as "en," for English.

Command Default

The welcome audio file downloaded with Cisco Unified CME B-ACD is used for the welcome prompt**.**

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param welcome-prompt** command.

Usage Guidelines

This command is used only with with a version of the Cisco Unified CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. The welcome greeting is the initial AA response to a caller. The default audio file used is en_bacd_welcome.au, which is is downloaded with Cisco CME B-ACD and announces, "Thank you for calling," and includes a two-second pause after the message.

The filename must be preceded by an underscore (_) character. In addition, it must not contain a language-code prefix, such as "en" for English. For example, for en_bacd_welcome.au, you must configure **welcome-prompt _bacd_welcome.au** instead of **welcome-prompt _en_bacd_welcome.au**.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example sets file name en_welcome.au as the welcome greeting for Cisco Unified CME B-ACD:

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
```

```
Router(config)# call application voice aa welcome-prompt _bacd_welcome_2.au
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice aa-name	Associates a Cisco CME B-ACD call queue script with a Cisco Unified CME B-ACD AA script
call application voice service-name	Associates a Cisco CME B-ACD AA script with a Cisco Unified CME B-ACD call queue script.

callback (voice emergency response settings)

To route an E911 callback to another number (for example, the company operator) if the callback cannot find the last 911 caller associated to the ELIN, use the **callback** command in voice emergency response settings configuration mode. This command is optional.

callback *number*
no callback

Syntax Description

number	Identifier of the E.164 default number to contact if a 911 callback fails.

Command Default

A callback number is not defined.

Command Modes

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to specify the default number to contact if a 911 callback cannot find the last 911 caller associated with an ELIN. If no default callback number is configured, and the expiry time is exceeded, the 911 operator may hear a reorder tone or be incorrectly routed.

Examples

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller's IP phone address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408-555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500.

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
```

Related Commands

Command	Description
elin	E.164 number used as the default ELIN if no matching ERL to the 911 caller's IP phone address is found.
expiry	Number of minutes a 911 call is associated to an ELIN in case of a callback from the 911 operator.
logging	Syslog informational message printed to the console every time an emergency call is made.

Command	Description
voice emergency response settings	Creates a tag for identifying settings for E911 behavior.

caller-id

To specify whether to pass the local caller ID or the original caller ID with calls from an extension in Cisco Unified CME that is using loopback, use the **callerid command in** ephone-dn configuration mode. To return to the default, use the **no** form of this command.

caller-id {**local** | **passthrough**}
no caller-id {**local** | **passthrough**}

Syntax Description

local	Local caller ID for redirected calls.
passthrough	Original caller ID. Default.

Command Default

Default is **passthrough**.

Command Modes

Ephone-dn configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ3	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command is valid only for ephone-dns that are being used for loopback.

This command with the **local** keyword is applied as follows:

- For transferred calls, caller ID is provided by the original caller-ID information source, such as from a separate loopback-dn that handles inbound calls or from a public switched telephone network interface.
- For forwarded calls, caller ID is provided by the original caller-ID information source or, for local IP phones, is extracted from the redirected information associated with the call.

This command with the **passthrough** keyword is applied as follows:

- For transferred calls, the caller ID is provided by the original caller-ID information that is obtained from the inbound side of the loopback-dn.
- For forwarded calls, the caller ID is provided by the original caller-ID information of the incoming call.

Examples

The following example selects local caller ID for redirected calls:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# loopback-dn 15 forward 4
Router(config-ephone-dn)# caller-id local
Router(config-ephone-dn)# no huntstop
```

Related Commands

Command	Description
loopback-dn	Creates a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP voice calls and supplementary services.

caller-id block (ephone-dn and ephone-dn-template)

To specify caller-ID blocking for outbound calls from a specific extension, use the **callerid block command in** ephone-dn or ephone-dn-template configuration mode. To disable caller-ID blocking for outbound calls, use the **no** form of this command.

caller-id block
no caller-id block

Syntax Description

This command has no arguments or keywords.

Command Default

Caller-ID display is not blocked on calls originating from a Cisco Unified IP phone.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command sets caller-ID blocking for outbound calls originating from a specific extension (ephone-dn). This command requests the far-end gateway device to block the display of the calling party information for calls received from the ephone-dn that is being configured. This command does not affect the ephone-dn calling party information display for inbound calls received by the ephone-dn.

If you want caller-ID name or number to be available on local calls but not on external calls, use the **clid strip name** command or the **clid strip** command in dial-peer configuration mode to remove caller-ID name or number from calls to VoIP. In this case, do not also use the **caller-id block** command, which blocks caller-ID information on all calls.

Note This command is not valid for ephone-dns that are being used for loopback.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example shows how to set caller-ID blocking for the directory number 5001:

```
Router(config)# ephone-dn 1
```

```
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# caller-id block
```

The following example uses an ephone-dn template to set caller-ID blocking for the directory number 5001:

```
Router(config)# ephone-dn-template 5
Router(config-ephone-dn-template)# caller-id block
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# ephone-dn-template 5
```

Related Commands

Command	Description
clid strip	Prevents display of caller-ID number on calls to VoIP.
clid strip name	Prevents display of caller-ID name on calls to VoIP.
ephone-dn-template (ephone-dn)	Applies ephone-dn template to an ephone dn.

caller-id block (voice register template)

> **Note** Effective with Cisco IOS Release 12.4(11)XJ, the **callerid block (voice register template)** command is not available in Cisco IOS software.

To enable caller-ID blocking for outbound calls from a specific SIP phone, use the **callerid block command in** voice register template configuration mode. To disable caller-ID blocking, use the **no** form of this command.

caller-id block
no caller-id block

Syntax Description

This command has no arguments or keywords.

Command Default

Caller ID blocking is disabled.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed.
12.4(15)T	Cisco Unified CME 4.1	This command was removed in Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command sets caller-ID blocking for outbound calls originating from any SIP phone that uses the specified template. This command requests the far-end gateway device to block the display of the calling party information for calls received from the specified SIP phone. This command does not affect the calling party information displayed for inbound calls received by the SIP phone. To apply a template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following example shows how to enable caller-ID blocking in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# caller-id block
```

Related Commands

Command	Description
anonymous block (voice register template)	Enables anonymous call blocking in a SIP phone template.
template (voice register pool)	Applies a template to a SIP phone.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

caller-id block code (telephony-service)

To set a code for a user to dial to block the display of caller ID on selected outgoing calls from Cisco IP phones, use the **caller-id block code** command in telephony-service configuration mode. To remove the code, use the **no** form of this command.

caller-id block code *code-string*
no caller-id block code

Syntax Description

code-string	Character string to dial to enable blocking of caller ID display on selected outgoing calls. The first character must be an asterisk (*) and the remaining characters must be digits. The string can contain a maximum of 16 characters.

Command Default

No caller-ID blocking code is defined.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Once the caller-ID blocking code has been defined using this command, phone users should enter the caller-ID blocking code before dialing any call on which they want their caller ID not to display.

Examples

The following example sets a caller-ID blocking code of *4321:

```
Router(config)# telephony-service
Router(config-telephony)# caller-id block code *4321
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

call-feature-uri

To specify the uniform resource identifier (URI) for soft keys on SIP phones registered to a Cisco Unified CME router, use the **call-feature-uri** command in voice register global configuration mode. To remove a URI association, use the **no** form of this command.

call-feature-uri {**cfwdall** | **gpickup** | **pickup**} *service-uri*
no call-feature-uri cfwdall {**cfwdall** | **gpickup** | **pickup**}

Syntax Description

cfwdall	Call Forward All (CfwdAll) soft key.
gpickup	Group Pickup (GPickUp) soft key.
pickup	Local Pickup (PickUp) soft key.
service-uri	URI that is requested when the specified soft key is pressed.

Command Default

No URI is associated with the soft key.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(22)YB	Cisco Unified CME 7.1	The **gpickup** and **pickup** keywords were added.
12.4(24)T	Cisco Unified CME 7.1	This command has been integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command updates the service URI for soft keys in the configuration file that is downloaded from the Cisco Unified CME router to the SIP phones during phone registration.

For Call Forward All, this URI and the call forward number is sent to Cisco Unified CME when a user enables Call Forward All from the phone using the CfwdAll soft key.

After you configure this command, restart the phone by using the **reset** or **restart** command.

This command is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

Examples

The following example shows how to specify the URI for the call forward all soft key:

```
Router(config)# voice register global
Router(config-register-global)# call-feature-uri cfwdall http://10.10.10.11/cfwdall
```

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP back-to-back user agent (B2BUA) so that all incoming calls are forwarded to another extension.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.
reset (voice register pool)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.
restart (voice register)	Performs a fast restart of one or all SIP phones associated with a Cisco Unified CME router.
service directed-pickup	Enables Directed Call Pickup and modifies the function of the PickUp and GPickUp soft keys.

call-forward

To globally apply dialplan-pattern expansion to redirecting numbers for extension numbers associated with SCCP IP phones in Cisco Unified CME, use the **call-forward system** command in telephony-service configuration mode. To disable the **call-forward system** command, use the **no** form of this command.

call-forward system redirecting-expanded
no call-forward system redirecting-expanded

Syntax Description

system	Call forward system parameter.
redirecting-expanded	Expand redirecting extensions to an E.164 number.

Command Default

The redirecting number is not expanded.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to apply dialplan-pattern expansion on a per-system basis to individual nonSIP redirecting numbers, including original called and last reroute numbers, in a Cisco Unified CME system.

When A calls B, and B forwards the call to C; B is the original called number and the last reroute number. If C then forwards or transfers the call to another number, C becomes the original called number and the last reroute number. The dial-plan pattern expansion is applied to both redirecting numbers. Once the number is expanded, it remains expanded during the entire call instance.

The dial-plan pattern to be matched must be configured using the **dialplan-pattern** command.

Examples

The following example shows how to create a dialplan-pattern for expanding calling numbers to an E.164 number and to also apply the expansion globally to redirecting numbers.

```
Router(config)# voice register global
Router(config-register-global)# dialplan-pattern 1 5105550... extension-length 5
Router(config-register-global)# call-forward system redirecting-expanded
```

Related Commands

Command	Description
dialplan-pattern	Create global prefix for expanding extension numbers of forward-to and transfer-to targets.
show telephony-service dial-peer	Displays dial peer information for extensions in a Cisco Unified CME system.

call-forward (voice register)

To globally apply dialplan-pattern expansion to redirecting numbers for extension numbers associated with SIP IP phones in Cisco Unified CME, use the **call-forward system** command in voice register global configuration mode. To disable the **call-forward system** command, use the **no** form of this command.

call-forward system redirecting-expanded
no call-forward system redirecting-expanded

Syntax Description

system	Call forward system parameter.
redirecting-expanded	Redirecting extension is to be expanded to an E.164 number.

Command Default

The redirecting number is not expanded.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9).

Usage Guidelines

Use this command to apply dialplan-pattern expansion on a per-system basis to individual SIP redirecting numbers, including original called and last reroute numbers, in Cisco Unified CME.

When A calls B, and B forwards the call to C; B is the original called number and the last reroute number. If C then forwards or transfers the call to another number, C becomes the original called number and the last reroute number. The dial-plan pattern expansion is applied to both redirecting numbers. Once the number is expanded, it remains expanded during the entire call instance.

This command supports call forward using B2BUA only.

The dial-plan pattern to be matched must be configured using the **dialplan-pattern** command.

Examples

The following example shows how to create a dialplan-pattern for expanding calling numbers of SIP phones to an E.164 number and to also apply the expansion globally to SIP redirecting numbers.

```
Router(config)# voice register global
Router(config-register-global)# dialplan-pattern 1 5105550... extension-length 5
Router(config-register-global)# call-forward system redirecting-expanded
```

Related Commands

Command	Description
dialplan-pattern (voice register)	Create global prefix for expanding extension numbers of forward-to and transfer-to targets if the target is an extension on a SIP phone.
show voice register dial-peer	Displays dial peer information for extensions in a Cisco Unified CME system.

call-forward all

To configure call forwarding so that all incoming calls to a directory number are forwarded to another directory number, use the **callforward all command in** ephone-dn or ephone-dn-template configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward all *directory-number*
no call-forward all

Syntax Description

directorynumber	Directory number to which calls are forwarded. Represents a fully qualified E.164 number.

Command Default

Call forwarding for all calls is not set.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The call forwarding mechanism applies to the individual directory number and cannot be configured for individual Cisco Unified IP phones.

Note The **callforward all** command takes precedence over the **call-forward busy** and **call-forward noan** commands.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example shows how to set call forwarding of all calls on directory number 5001 to directory number 5005. All incoming calls destined for extension 5001 are forwarded to another Cisco IP phone with the extension number 5005:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# call-forward
 all 5005
```

The following example uses an ephone-dn template to forward all calls for extension 5001 to extension 5005.

```
Router(config)# ephone-dn-template 3
Router(config-ephone-dn-template)# call-forward all 5005
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# ephone-dn-template 3
```

Related Commands

Command	Description
callforward busy	Configures call forwarding to another number when a Cisco Unified IP phone is busy.
callforward noan	Configures call forwarding to another number when no answer is received from a Cisco Unified IP phone.
ephonedn-template (ephone-dn)	Applies template to ephone-dn.

call-forward b2bua all

To enable call forwarding for a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA) so that all incoming calls are forwarded to another extension, use the **callforward b2bua all command in** voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua all *directory-number*
no call-forward b2bua all

Syntax Description

directorynumber	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.

Command Default

Feature is disabled.

Command Modes

Voice register dn configuration (config-register-dn)
Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed from voice register pool configuration mode for Cisco Unified CME only.
12.4(15)T	Cisco Unified CME 4.1	Command with modifications was integrated into Cisco IOS release 12.4(15)T.

Usage Guidelines

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

If this command is configured in both the voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

The **call-forward b2bua all** command takes precedence over the **call-forward b2bua busy** and **call-forward b2bua noan** commands.

Note

This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

Examples

Cisco Unified CME and Cisco Unified SIP SRST

The following example shows how to forward all incoming calls to extension 5001 on directory number 4, to extension 5005.

```
Router(config)# voice register dn 4
Router(config-register-dn)# number 5001
Router(config-register-dn)# call-forward b2bua all 5005
```

Cisco Unified SIP SRST

The following example shows how to forward all incoming calls for extension 5001 on pool number 4, to extension 5005.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua all 5005
```

Related Commands

Command	Description
call-forward b2bua busy	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.
call-forward b2bua mailbox	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-forward b2bua noan	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.

call-forward b2bua busy

To enable call forwarding for a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA) so that incoming calls to a busy extension are forwarded to another extension, use the **callforward b2bua busy command in** voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua busy *directory-number*
no call-forward b2bua busy

Syntax Description

directorynumber	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.

Command Default

Feature is disabled.

Command Modes

Voice register dn configuration (config-register-dn)
Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed from voice register pool configuration mode for Cisco Unified CME only.
12.4(15)T	Cisco Unified CME 4.1	This command with modifications was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST that is off-hook. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

In Cisco Unified CME, call forward busy is also invoked when a call arrives for a destination that is configured but unregistered. A destination is considered to be configured if its number is listed under the voice register dn configuration.

If this command is configured in both voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

The **call-forward b2bua all** command takes precedence over the **call-forward b2bua busy** and **call-forward b2bua noan** commands.

Note This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

Cisco Unified CME and Cisco Unified SIP SRST

The following example shows how to forward all incoming calls to extension 5001 on directory number 4 to extension 5005 when extension 5001 is busy.

```
Router(config)# voice register dn 4
Router(config-register-dn)# number 5001
Router(config-register-dn)# call-forward b2bua busy 5005
```

Cisco Unified SIP SRST

The following example shows how to forward calls from extension 5001 in pool 4 to extension 5005 when extension 5001 is busy.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua busy 5005
```

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
call-forward b2bua mailbox	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-forward b2bua noan	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.

call-forward b2bua mailbox

To control the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange, use the **callforward b2bua mailbox command in** voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua mailbox *directory-number*
no call-forward b2bua mailbox

Syntax Description

directorynumber	Telephone number to which calls are forwarded when the forwarded destination is busy or does not answer. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.

Command Default

Feature is disabled.

Command Modes

Voice register dn configuration (config-register-dn)
Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed from voice register pool configuration mode for Cisco Unified CME only.
12.4(15)T	Cisco Unified CME 4.1	This command with modifications was integrated into Cisco IOS Release 12.4(15)T

Usage Guidelines

This command is used to denote the voice-mail box to use at the end of a chain of call forwards to busy or no answer destinations. It can be used to forward calls to a voice-mail box that has a different number than the forwarding extension, such as a shared voice-mail box.

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

If this command is configured in both the voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

This command is used in conjunction with the **call-forward b2bua all**, **call-forward b2bua busy**, and **call-forward b2bua noan** commands.

Note This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

Cisco Unified CME and Cisco Unified SIP SRST

The following example shows how to forward all incoming calls to extension 5005 if an incoming call is forwarded to extension 5001, and extension 5001 is busy or does not answer.

```
Router(config)# voice register dn 4
Router(config-register-dn)# number 5001
Router(config-register-dn)# call-forward b2bua mailbox 5005
```

Cisco Unified SIP SRST

The following example shows how to forward calls to extension 5005 if an incoming call is forwarded to extension 5001 on pool number 4, and extension 5001 is busy or does not answer.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua mailbox 5005
```

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
call-forward b2bua busy	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.
call-forward b2bua noan	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.
call-forward b2bua unreachable	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that is not registered in Cisco Unified CME are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.
number (voice register dn)	Associates an extension number with a voice register dn.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
voice register pool	Enters voice register pool configuration mode for SIP phones.

call-forward b2bua night-service

To automatically forward calls to another number during night-service hours, use the **call-forward b2bua night-service** command in voice register dn configuration mode. To remove the code, use the **no** form of this command.

call-forward b2bua night-service *target-number*
no call-forward b2bua night-service

Syntax Description

target-number	Phone number to which calls are forwarded.

Command Default

Calls are not forwarded during night-service hours.

Command Modes

Voice register dn configuration: (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

You need to configure the **night-service bell** command under voice register dn. Night-service hours are defined using the **night-service date** and **night-service day** commands.

A voice register dn can have all other types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding destination defined in its target-number argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

- call forward night-service (only during night service hours)
- call forward all
- call forward busy and call forward no answer

Examples

The following example defines a call forward night-service configuration under voice register dn :

```
Router(config)# voice register dn tag
Router(config-register-dn)# call-forward b2bua night-service
```

Related Commands

	Description
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

call-forward b2bua noan

To enable call forwarding for a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA) so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension, use the **callforward b2bua noan command in** voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua noan *directory-number* **timeout** *seconds*
no call-forward b2bua noan

Syntax Description

directorynumber	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.
timeout *seconds*	Number of seconds that a call can ring with no answer before the call is forwarded to another extension. Range is 3 to 60000. Default is 20.

Command Default

Feature is disabled.

Command Modes

Voice register dn configuration (config-register-dn)
Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed from voice register pool configuration mode for Cisco Unified CME only.
12.4(15)T	Cisco Unified CME 4.1	This command with modifications was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST that remains unanswered after a specified length of time. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

If this command is configured in both the voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

The **call-forward b2bua all** command takes precedence over the **call-forward b2bua busy** and **call-forward b2bua noan** commands.

Note This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

Cisco Unified CME and Cisco Unified SIP SRST

The following example shows how to forward calls to extension 5005 when extension 5001 is unanswered. The timeout before the call is forwarded to extension 5005 is 10 seconds.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua noan 5005 timeout 10
```

Cisco Unified SIP SRST

The following example shows how to forward calls to extension 5005 when extension 5001 on pool number 4 is unanswered. The timeout before the call is forwarded to extension 5005 is 10 seconds.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua noan 5005 timeout 10
```

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
call-forward b2bua busy	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.
call-forward b2bua mailbox	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.

call-forward b2bua unreachable

Note Effective with Cisco IOS Release 12.4(11)XJ, the **callforward b2bua unreachable** command is not available in Cisco IOS software.

To forward calls to a phone that is not registered to Cisco Unified CME, use the **callforward b2bua unreachable command in** voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua unreachable *directory-number*
no call-forward b2bua unreachable

Syntax Description

directorynumber	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number.

Command Default

Feature is disabled

Command Modes

Voice register dn configuration (config-register-dn)
Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed.
12.4(15)T	Cisco Unified CME 4.1	This command was removed in Cisco IOS Release 12.4(15)T.

Usage Guidelines

Call forward unreachable is triggered when a call arrives for a destination that is configured but unregistered with Cisco CME. A destination is considered to be configured if its number is listed under the voice register pool or voice register dn configurations.

If call forward unreachable is not configured for a pool or directory number (DN) register, any calls that match the numbers in that pool or DN register will use call forward busy instead.

We recommend that you do not use this command with hunt groups. If the command is used, consider removing the phone from any assigned hunt groups, unless you want to forward calls to all phones in the hunt group.

Examples

The following example shows how to forward calls to extension 5005 when extension 5001 on directory number 4 is unreachable, either because it is unplugged or the network between the Cisco router and the extension is nonfunctional. The timeout before the call is forwarded to extension 5005 is 10 seconds.

```
Router(config)# voice register pool 4
Router(config-register-dn)# number 5001
Router(config-register-dn)# call-forward b2bua unreachable 5005 timeout 10
```

Related Commands

Command	Description
call-forward b2bua all (voice register dn and voice register pool)	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
call-forward b2bua busy (voice register dn and voice register pool)	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.
call-forward b2bua mailbox (voice register dn and voice register pool)	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-forward b2bua noan (voice register dn and voice register pool)	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.
number (voice register dn)	Associates an extension number with a voice register dn.

call-forward busy

To configure call forwarding so that incoming calls to a busy extension (ephone-dn) are forwarded to another extension, use the **callforward busy command in** ephone - dn or ephone-dn-template configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward busy *target-number* [{**primary** | **secondary**}] [**dialplan-pattern**]
no call-forward busy

Syntax Description

target-number	Phone number to which calls are forwarded.
primary	(Optional) Call forwarding is selectively applied only to the dial peer created for the primary number for this ephone-dn.
secondary	(Optional) Call forwarding is selectively applied only to the dial peer created for the secondary number for this ephone-dn.
dialplan-pattern	(Optional) Call forwarding is selectively applied only to dial peers created for this ephone-dn by the dial-plan pattern.

Command Default

Call forwarding for a busy extension is not enabled.

Command Modes

Ephone-dn configuration (config-dn-ephone)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	Cisco Unified CME 4.0	The **primary**, **secondary**, and **dialplan-pattern** keywords were added, and this command was made available in ephone-dn-template configuration mode.
12.4(11)T	Cisco Unified CME 4.0	This command with the **primary**, **secondary**, and **dialplan-pattern** keywords added, and this command in ephone-dn-template configuration mode was integrated into Cisco IOS 12.4(11)T.

Usage Guidelines

The call forwarding mechanism is applied to an individual extension (ephone-dn) and is not applied to the phone on which the extension appears.

Normally, call forwarding is applied to all dial peers that are created by the ephone-dn. An ephone-dn can create up to four dial peers:

- A dial peer for the primary number
- A dial peer for the secondary number
- A dial peer for the primary number as expanded by the **dialplan-pattern** command
- A dial peer for the secondary number as expanded by the **dialplan-pattern** command

The **primary**, **secondary**, and **dialplan-pattern** keywords allow you to apply call forwarding selectively to one or more dial peers based on the exact called number that was used to route the call to the ephone-dn. If none of the optional keywords is used, call forwarding applies to all dial-peers that are defined for the ephone-dn.

An ephone-dn can have all four types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding target defined in its *target-number* argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

1. call forward night service
2. call forward all
3. call forward busy and call forward no answer

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example forwards all calls for the ephone-dn 2345 when it is busy.

```
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# call-forward
 busy 2000
```

The following example uses an ephone-dn template to forward calls for extension 2345 when it is busy.

```
Router(config)# ephone-dn-template 6
Router(config-ephone-dn-template)# call-forward
 busy 2000
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# ephone-dn-template 6
```

The following example creates a dial-plan pattern to expand extension numbers into E.164 numbers. It then sets call forwarding of incoming calls to directory number 5005. In this example, call forwarding on busy is applied only when callers dial the primary number for this ephone-dn, 5001.

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855501.. extension-length 4 extension-pattern
 50..
Router(config-telephony)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001 secondary 5002
Router(config-ephone-dn)# call-forward
 busy 5005 primary
```

Related Commands

Command	Description
callforward all	Configures call forwarding for all incoming calls to an ephone-dn.

Command	Description
call-forward night-service	Configures call forwarding for all incoming calls to an ephone-dn during the hours defined for night service.
callforward noan	Configures call forwarding to another number when no answer is received from an ephone-dn.
ephone-dn-template (ephone-dn)	Applies template to ephone-dn.

call-forward max-length

To restrict the number of digits that can be entered using the CfwdALL soft key on an IP phone, use the **callforward max-length command in** ephone-dn or ephone-dn-template configuration mode. To remove a restriction on the number of digits that can be entered, use the **no** form of this command.

call-forward max-length *length*
no call-forward max-length

Syntax Description

length	Number of digits that can be entered using the CfwdAll soft key on an IP phone.

Command Default

There is no restriction on the number of digits that can be entered.

Command Modes

Ephone-dn configuration (config-dn-ephone)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(7)T	Cisco CME 3.1	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(11)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(11)T.

Usage Guidelines

This command can be used to prevent a phone user from using the CfwdALL soft key on an IP phone to forward calls to numbers that will incur toll charges when they receive forwarded calls.

If the *length* argument is set to 0, the CfwdALL soft key is completely disabled. If the ephone-dn associated with the first line button has an active call forward number when this command is used to set the *length* argument to 0, the CfwdALL soft key will be disabled after the next phone restart.

The restriction created by this command does not apply to destinations that are entered using the Cisco IOS command-line interface (CLI) or the Cisco Unified CME GUI.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example restricts the number of digits that a phone user can enter using the CfwdALL soft key to four. In this example, extensions in the phone user’s Cisco Unified CME system have four digits, so that means the user can use the IP phone to forward all calls to any extension in the system, but not to any number outside the system.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# call-forward
 max-length 4
```

The following example uses an ephone-dn-template to restrict the number of digits that a phone user can enter using the CfwdALL soft key to four.

```
Router(config)# ephone-dn-template 4
Router(config-ephone-dn-template)# call-forward
 max-length 4
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# ephone-dn-template 4
```

Related Commands

Command	Description
call-forward all	Configures call forwarding for all incoming calls on one of the lines of a Cisco Unified IP phone.
ephone-dn-template (ephone-dn)	Applies an ephone-dn template to an ephone-dn.

call-forward night-service

To automatically forward calls to another number during night-service hours, use the **call-forward night-service** command in ephone-dn or ephone-dn-template configuration mode. To disable automatic call forwarding during night service, use the **no** form of this command.

call-forward night-service *target-number*
no call-forward night-service

Syntax Description

target-number	Phone number to which calls are forwarded.

Command Default

Calls are not forwarded during night-service hours.

Command Modes

Ephone-dn configuration (config-dn-ephone)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(11)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(11)T.

Usage Guidelines

You must also configure the **night-service bell** command for this ephone-dn.

Night-service hours are defined using the **night-service date** and **night-service day** commands.

An ephone-dn can have all four types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding destination defined in its *target-number* argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

1. call forward night-service
2. call forward all
3. call forward busy and call forward no answer

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example establishes night-service hours from 1 p.m. Saturday until 8 a.m. Monday. During that time, calls to extension 1000 (ephone-dn 1) are forwarded to extension 2346. Note that the **night-service bell** command has also been used for ephone-dn 1.

```
telephony-service
 night-service day sat 13:00 12:00
 night-service day sun 12:00 08:00
 night-service code *1234
!
ephone-dn 1
 number 1000
```

```
 night-service bell
 call-forward night-service 2346
!
ephone-dn 2
 number 2346
ephone 12
 button 1:1
ephone 13
 button 1:2
```

The following example uses an ephone-dn template to apply call forwarding for extension 2876 during the night service hours established in the previous example.

```
ephone-dn-template 2
 call-forward night-service 2346
ephone-dn 25
 number 2876
 ephone-dn-template 2
```

Related Commands

Command	Description
callforward all	Configures call forwarding for all incoming calls to an ephone-dn.
callforward busy	Configures call forwarding to another number when an ephone-dn is busy.
callforward noan	Configures call forwarding to another number when no answer is received from an ephone-dn.
night-service bell (ephone-dn)	Marks an ephone-dn for night-service treatment.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

call-forward noan

To configure call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another number, use the **callforward noan command in** ephone - dn or ephone-dn-template configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward noan *target-number* **timeout** *seconds* [{**primary** | **secondary**}] [**dialplan-pattern**]
no call-forward noan

Syntax Description

target-number	Phone number to which calls are forwarded.
timeout *seconds*	Sets the duration that a call can ring with no answer before the call is forwarded to the target number. Range is from 3 to 60000. There is no default value.
primary	(Optional) Call forwarding is selectively applied only to the dial peer created for the primary number for this ephone-dn.
secondary	(Optional) Call forwarding is selectively applied only to the dial peer created for the secondary number for this ephone-dn.
dialplan-pattern	(Optional) Call forwarding is selectively applied only to dial peers created for this ephone-dn by the dial-plan pattern.

Command Default

Call forwarding for an extension that does not answer is not enabled.

Command Modes

Ephone-dn configuration (config-dn-ephone)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	Cisco Unified CME 4.0	The **primary**, **secondary**, and **dialplan-pattern** keywords were added, and this command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command with modifications was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The call forwarding mechanism is applied to an individual extension (ephone-dn) and is not applied to the phone on which the extension appears.

Normally, call forwarding is applied to all dial peers that are created by the ephone-dn. An ephone-dn can create up to four dial peers:

- A dial peer for the primary number
- A dial peer for the secondary number
- A dial peer for the primary number as expanded by the **dialplan-pattern** command

- A dial peer for the secondary number as expanded by the **dialplan-pattern** command

The **primary**, **secondary**, and **dialplan-pattern** keywords allow you to apply call forwarding selectively to one or more dial peers based on the exact called number that was used to route the call to the ephone-dn. If none of the optional keywords is used, call forwarding applies to all dial-peers that are defined for the ephone-dn.

An ephone-dn can have all four types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding target defined in its *target-number* argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

1. call forward night service
2. call forward all
3. call forward busy and call forward no answer

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example forwards calls for the ephone-dn 2345 when it does not answer.

```
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# call-forward
 busy 2000
The following example uses an ephone-dn-template to forward calls for the ephone-dn 2345
when it does not answer.
Router(config)# ephone-dn-template 8
Router(config-ephone-dn-template)# call-forward
 busy 2000
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# ephone-dn-template 8
```

The following example creates a dial-plan pattern to expand extension numbers into E.164 numbers. It then sets call forwarding of incoming calls to directory number 5005. In this example, call forwarding on no answer is applied only when callers dial the primary number for this ephone-dn, 5001.

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855501.. extension-length 4 extension-pattern
 50..
Router(config-telephony)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001 secondary 5002
Router(config-ephone-dn)# call-forward
 noan 5005 primary
```

Related Commands

Command	Description
callforward all	Configures call forwarding for all incoming calls for an ephone-dn.

Command	Description
callforward busy	Configures call forwarding to another number when an ephone-dn is busy.
call-forward night-service	Configures call forwarding for all incoming calls to an ephone-dn during the hours defined for night service.
ephone-dn-template (ephone-dn)	Applies an ephone-dn-template to an ephone-dn.

call-forward pattern

To specify a pattern for calling - party numbers that are able to support the ITU-T H.450.3 standard for call forwarding, use the **callforward pattern** command in telephony-service configuration mode. To remove the pattern, use the **no** form of this command.

call-forward pattern *pattern*
no call-forward pattern *pattern*

Syntax Description

pattern	String that consists of one or more digits and wildcard markers or dots (.) to define a specific pattern. Calling parties that match a defined pattern use the H.450.3 standard if they are forwarded. A pattern of .T specifies the H.450.3 forwarding standard for all incoming calls.

Command Default

No call-forward pattern is defined.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco CME 2.1	This command was introduced.
12.2(15)T	Cisco CME 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Use this command with Cisco IOS Telephony Services (ITS) V2.1, Cisco CallManager Express 3.0, or a later version.

When H.450.3 call forwarding is selected, the router must be configured with a Tool Command Language (Tcl) script that supports the H.450.3 protocol. The Tcl script is loaded on the router by using the **call application voice** command.

The pattern match in this command is against the phone number of the calling party. When an extension number has forwarded its calls and an incoming call is received for that number, the router sends an H.450.3 response back to the original calling party to request that the call be placed again using the forward-to destination.

Calling numbers that do not match the patterns defined using this command are forwarded using Cisco-proprietary call forwarding for backward compatibility.

Examples

The following example specifies that all 4-digit directory numbers that begin with 4 should use the H.450.3 standard whenever they are forwarded:

```
Router(config)# telephony-service
Router(config-telephony)# call-forward pattern 4...
```

The following example forwards all calls that support the H.450.3 standard:

```
Router(config)# telephony-service
Router(config-telephony)# call-forward pattern .T
```

Related Commands

Command	Description
call application voice	Defines an application, indicates the location of the corresponding Tcl files that implement the application, and loads the selected Tcl script.

calling-number local

To replace a calling-party number and name with the forwarding-party number and name (the local number and name) in calls forwarded using local hairpin call routing, use the **calling-number local** command in telephony-service configuration mode. To reset to the default, use the **no** form of this command.

calling-number local [**secondary**]
no calling-number local

Syntax Description

secondary	(Optional) Uses the secondary number associated with the forwarding party instead of the primary number. The primary number is the default if this keyword is not used.

Command Default

Calling-party numbers and names are used in forwarded calls.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ3	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(15)ZJ4	Cisco CME 3.0	The **secondary** keyword was introduced.
12.3(14)T	Cisco CME 3.3	Support was added to the default IOS voice application framework and dependency on the TCL script was removed.

Usage Guidelines

In Cisco CME 3.2 and earlier versions, this command is used with the Tool Command Language (Tcl) script app-h450-transfer.2.0.0.7 or a later version.

In Cisco CME 3.3 and later versions, this command can be used without the TCL script because the functionality is integrated into the default IOS voice application framework.

If the ephone-dn used by a forwarding party has a secondary number in addition to its primary number and neither number is registered with the gatekeeper, the primary number is the number that appears as the calling number on hairpin-forwarded calls when the **calling-number local** command is used. If only one of the numbers is registered with the gatekeeper, the registered number is the number that appears as the calling number. If both numbers are registered with the gatekeeper, the primary number is the number that appears as the calling number.

If the ephone-dn used by a forwarding party has a secondary number in addition to its primary number and the **calling-number local secondary** command is used, the secondary number is the number that appears as the calling number on hairpin-forwarded calls if both numbers are registered with the gatekeeper or if both numbers are not registered. If only one number is configured to register with the gatekeeper, the number that is registered appears as the calling number.

Examples

The following example specifies use of the name and number of the local forwarding party in hairpin-forwarded calls:

```
Router(config)# telephony-service
```

```
Router(config-telephony)# calling-number local
```

The following examples demonstrate the use of the **calling-number local** command without the **secondary** keyword.

- The calling number for hairpin calls forwarded from ephone-dn 1 is 1234 in the following example:

```
calling-number local
ephone-dn 1
 number 1234 secondary 4321 no-reg
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 4321 in the following example:

```
calling-number local
ephone-dn 1
 number 1234 secondary 4321 no-reg primary
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 1234 in the following example:

```
calling-number local
ephone-dn 1
 number 1234 secondary 4321 no-reg both
```

or

```
 number 1234 secondary 4321
```

The following examples demonstrate the use of the **calling-number local secondary** command.

- The calling number for hairpin calls forwarded from ephone-dn 1 is 1234 in the following example:

```
calling-number local secondary
ephone-dn 1
 number 1234 secondary 4321 no-reg
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 4321 in the following example:

```
calling-number local secondary
ephone-dn 1
 number 1234 secondary 4321 no-reg primary
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 4321 in the following example:

```
calling-number local secondary
ephone-dn 1
 number 1234 secondary 4321 no-reg both
```

or

```
 number 1234 secondary 4321
```

calling-number local (voice register global)

To replace a calling-party number and name with the forwarding-party number and name (the local number and name) in calls forwarded using local hairpin call routing, use the **calling-number local** command in voice register global configuration mode. To reset to the default, use the **no** form of this command.

calling-number local
no calling-number local

Syntax Description

This command has no arguments or keywords.

Command Default

Calling-party numbers and names are used in forwarded calls. The command is disabled by default.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.6.1	Unified CME 12.0	This command was introduced.

Usage Guidelines

Use the CLI Command **calling-number local** in voice register global configuration mode so that the number and name of the forwarding party appears as the calling number on hairpin-forwarded calls. Once **calling-number local** is configured under voice register global, the calls forwarded from local SIP phones will have the calling-number and name of the last forwarded party.

Examples

The following example specifies use of the name and number of the local forwarding party in hairpin-forwarded calls:

```
Router(config)# voice register global
Router(config-register-global)# calling-number local
```

The following examples demonstrate the use of the **calling-number local** command.

- The calling number for hairpin calls forwarded from voice register dn 1 is 1234 in the following example:

```
voice register global
calling-number local
..
voice register dn 1
 name Phone 1
 number 1234
```

callqueue-display

To configure call waiting notification display on the agent phone as continuous, periodic, or off, use the **callqueue display** command. To set the call waiting notification display to the default state of periodic (for voice hunt group) and continuous (for ephone hunt group), use the **default** form of this command.

callqueue display [{**continuous** | **periodic** | **off**}]
default callqueue display

Command Default

Call waiting notification is set to periodic for phones in voice hunt group, and to continuous for phones in ephone hunt group. The no form of this command also sets the call waiting display to default state.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Voice hunt group configuration (config-voice-hunt-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

The callqueue display command is valid for both voice hunt group as well as ephone hunt group.

Examples

The following example shows how to set call waiting notification display to periodic in an ephone hunt group:

```
Router(config)# ephone-hunt 1
Router(config-ephone-hunt)# call
Router(config-ephone-hunt)# callqueue
Router(config-ephone-hunt)# callqueue display
Router(config-ephone-hunt)# callqueue display periodic
```

The following example shows how to set call waiting notification display to continuous in a voice hunt group:

```
Router(config)# voice hunt-group 1
Router(config-voice-hunt-group)# callqueue display
Router(config-voice-hunt-group)# callqueue display continuous
```

call-park system

To define system parameters for the Call Park feature, use the **call-park system** command in telephony-service configuration mode. To reset to the default, use the **no** form of this command.

call-park system {**application** | **redirect**}
no call-park system {**application** | **redirect**}

Syntax Description

application	Enables Call Park and Directed Call Park for SCCP and SIP phones.
redirect	H.323 and SIP calls use H.450 or the SIP Refer method of call forwarding or transfer to park calls and to pick up calls from park.

Command Default

H.323 and SIP calls use hairpin call forwarding or transfer to park calls and to pick up calls from park.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(22)YB	Cisco Unified CME 7.1	The **application** keyword and support for SIP phones was added.
12.4(24)T	Cisco Unified CME 7.1	This command has been integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

The **application** keyword selects the enhanced Call Park method supported in Cisco Unified CME 7.1 and later versions for SCCP and SIP phones.

Examples

The following example specifies that H.323 and SIP calls will use the H.450 or SIP Refer method of call forwarding or transfer to park calls and pick up calls from park:

```
Router(config)# telephony-service
Router(config-telephony)# call-park system redirect
```

Related Commands

Command	Description
park reservation-group	Assigns a call-park reservation group to a phone.
park-slot	Creates a floating extension at which calls can be temporarily parked.

call-waiting (voice register pool)

To enable call-waiting option on a SIP phone, use the **call-waiting** command in voice register pool configuration mode. To disable call waiting, use the **no** form of this command.

call-waiting
no call-waiting

Syntax Description

This command has no arguments or keywords.

Command Default

Feature is enabled.

Command Modes

Voice register pool configuration (call-waiting)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

The call waiting feature is enabled by default on SIP phones. To disable call waiting, use the **no call-waiting** command.

Examples

The following example shows how to disable call waiting on SIP phone 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# no call-waiting
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode for SIP phones.

call-waiting beep

To allow call-waiting beeps to be accepted by or generated from an ephone-dn, use the **call-waiting beep** command in ephone-dn or ephone-dn-template configuration mode. To disable the acceptance and generation of call-waiting beeps by an ephone-dn, use the **no** form of this command.

call-waiting beep [{**accept** | **generate**}]
no call-waiting beep [{**accept** | **generate**}]

Syntax Description

accept	(Optional) Allows call-waiting beeps to be accepted by an ephone-dn.
generate	(Optional) Allows call-waiting beeps to be generated by an ephone-dn.

Command Default

Call-waiting beeps are accepted and generated.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **call-waiting beep** command must be used with the **ephone-dn** command. The **call-waiting beep** command is used like a toggle and can be switched on and off for each ephone-dn.

A beep can be heard only if both sending and receiving ephone-dns are configured to accept call-waiting beeps.

To display how call-waiting beeps are configured, use the **show running-config** command in the privileged EXEC configuration mode. If the **no call-waiting beep generate** and **no call-waiting beep accept** commands are configured, the **show running-config** output will display the **no call-waiting beep** command.

If you configure a button to have a silent ring using the **s** option of the **button** command, you will not hear a call-waiting beep regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting beep.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example configures ephone-dn 1 and ephone-dn 2 not to accept and not to generate call-waiting beeps:

```
Router(config)# ephone-dn 1
```

```
Router(config-ephone-dn)# number 2588
Router(config-ephone-dn)# no call-waiting beep accept
Router(config-ephone-dn)# no call-waiting beep generate
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 2589
Router(config-ephone-dn)# no call-waiting beep accept
Router(config-ephone-dn)# no call-waiting beep generate
Router(config-ephone-dn)# exit
```

The following example uses an ephone-dn template to set the same attributes as in the previous example:

```
Router(config)# ephone-dn-template 5
Router(config-ephone-dn-template)# no call-waiting beep accept
Router(config-ephone-dn-template)# no call-waiting beep generate
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2588
Router(config-ephone-dn)# ephone-dn-template 5
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 2589
Router(config-ephone-dn)# ephone-dn-template 5
Router(config-ephone-dn)# exit
```

The following example configures ephone-dn 1 and ephone-dn 2 to switch back to accept call-waiting beeps. Ephone-dn 1 and ephone-dn 2 now accept but do not generate call-waiting beeps.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# call-waiting beep accept
Router(config)# ephone-dn 2
Router(config-ephone-dn)# call-waiting beep accept
```

Related Commands

Command	Description
show running-config	Displays the contents of the currently running configuration file or the configuration for a specific interface, or map class information.
ephone-dn-template (ephone-dn)	Applies a template to an ephone-dn.

call-waiting ring

To allow an ephone-dn to use a ring sound for call-waiting notification, use the **call-waiting ring** command in ephone-dn or ephone-dn-template configuration mode. To disable the ring notification, use the **no** form of this command.

call-waiting ring
no call-waiting ring

Syntax Description

This command has no arguments or keywords.

Command Default

The ephone-dn accepts call waiting and uses beeps for notification.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

To use a ring sound for call-waiting notification on an ephone-dn, you must ensure that the ephone-dn will accept secondary calls while it is connected to another line. The acceptance of call waiting is the default ephone-dn behavior. However, the **no call-waiting beep accept** command can change this default so an ephone-dn does not accept call waiting. This command must be removed for ringing notification to work.

The **call-waiting ring** command will automatically disable a call-waiting beep configuration.

If you configure a button to have a silent ring using the **s** option of the **button** command, you will not hear a call-waiting ring regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting ring.

Note The call-waiting ring option cannot be used on the Cisco Unified IP Phone 7902, Cisco Unified IP Phone 7905, or Cisco Unified IP Phone 7912. Do not use the **call-waiting ring** command for ephone-dns associated with these types of phones.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example configures ephone-dn 1 and ephone-dn 2 to use ringing for their call-waiting notification:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# call-waiting ring
Router(config)# ephone-dn 2
Router(config-ephone-dn)# no call-waiting ring
```

The following example uses an ephone-dn template to set the same attributes as in the previous example:

```
Router(config)# ephone-dn-template 9
Router(config-ephone-dn-template)# call-waiting ring
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn-template 10
Router(config-ephone-dn-template)# no
call-waiting ring
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# ephone-dn-template 9
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 2
Router(config-ephone-dn)# ephone-dn-template 10
Router(config-ephone-dn)# exit
```

Related Commands

Command	Description
call-waiting beep	Allows call-waiting beeps to be accepted by or generated from an ephone-dn.
ephone-dn-template (ephone-dn)	Applies template to ephone-dn.

camera

To enable USB camera capability on Cisco Unified IP Phones 9951 and 9971, use the **camera** command in voice register global, voice register template, and voice register pool configuration modes. To disable video capabilities on Cisco Unified IP Phones 9951 and 9971, use the **no** form of this command.

camera
no camera

Syntax Description

This command has no arguments or keywords.

Command Default

USB camera capability is disabled on Cisco Unified IP Phones 9951 and 9971.

Command Modes

Voice register global
Voice register template
Voice register pool

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.

Usage Guidelines

Use this command to enable USB camera capability on Cisco Unified IP Phones 9951 and 9971. You need to create profile and apply-config or restart to the phone to enable the video capability on phones.

Examples

The following example shows camera command configured in voice register global:

```
Router#show run
!
!
!
voice service voip
 allow-connections sip to sip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
!
voice register global
 mode cme
 bandwidth video tias-modifier 244 negotiate end-to-end
 max-pool 10
 camera
voice register template  10
!
!
```

The following example shows video and camera commands configured under voice register pool 5, you can also configure both camera and video commands under voice register template:

```
Router#show run
!
!
voice service voip
 allow-connections sip to sip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```

```
!
!
voice register global
 mode cme
 bandwidth video tias-modifier 244 negotiate end-to-end
 max-pool 10
!
voice register pool  1
 id mac 1111.1111.1111
!
voice register pool  4
!
voice register pool  5
 logout-profile 58
 id mac 0009.A3D4.1234
 camera
!
```

Related Commands

Command	Description
apply-config	Allows to dynamically apply the phone configuration on Cisco Unified SIP IP phones 8961, 9951, and 9971,

capf-auth-str

To define a string of digits that a user enters at the phone for CAPF authentication, use the **capf-auth-str** command in ephone configuration mode. To return to the default, use the **no** form of this command.

capf-auth-str *digit-string*
no capf-auth-str

Syntax Description

digit-string	String of digits that a phone user enters at the phone for CAPF authentication.

Command Default

No authentication string exists for the phone.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to create or remove an authentication string (Personal Identification Number or PIN) for the specified secure ephone. Use this command if the **auth-string** keyword is specified in the **auth-mode** command. Once you specify a CAPF authentication string, it becomes part of the ephone configuration. This value can also be set globally or per ephone using the **auth-string** command in CAPF configuration mode.

Use the **show capf-server auth-str** command to display configured authentication strings.

When a phone is configured for a certificate upgrade that requires auth-string authentication, the CAPF initiation needs to be performed manually by the phone user using the following steps:

1. Press the Settings button.
2. If the configuration is locked, press **# (asterisk, asterisk, pound sign) to unlock it.
3. Scroll down the menu and select Security Configuration.
4. Scroll down the next menu to LSC and press the Update soft key.
5. When prompted for the authentication string, enter the string provided by the system administrator.

Examples

The following example specifies the type of authentication for ephone 392 is an authentication string that is entered from the phone, and then defines the string as 38593.

```
ephone 392
 button 1:23 2:24 3:25
 device-security-mode authenticated
 cert-oper upgrade auth-mode auth-string
 capf-auto-str 38593
```

Related Commands

Command	Description
auth-mode	Specifies the type of authentication to use during CAPF sessions.
auth-string	Generates or removes authentication strings for one or all secure ephones.
show capf-server	Displays configuration and session information for the CAPF server.

capf-server

To enter CAPF-server configuration mode to set CAPF server parameters, use the **capf-server** command in global configuration mode. To remove the CAPF server configuration, use the **no** form of this command.

capf-server
no capf-server

Syntax Description

This command has no keywords or arguments.

Command Default

No CAPF server configuration is present.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example sets parameters for the CAPF server:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

cert-enroll-trustpoint

To enroll the CAPF with the CA or RA, use the **cert-enroll-trustpoint** command in CAPF-server configuration mode. To remove an enrollment, use the **no** form of this command.

cert-enroll-trustpoint *ca-label* **password** {**0** | **1**} *password-string*
no cert-enroll-trustpoint

Syntax Description

ca-label	PKI trustpoint label for the CA or for the RA if an RA is being used.
password	Values that follow apply to the password.
0 | **1**	Encryption status of the password string that follows. • **0**—Encrypted. • **1**—Clear text. **Note** This option refers to the way that you want the password to appear in show command output and not to the way that you enter the password in this command.
password-string	Alphanumeric challenge password that is required for certificate enrollment.

Command Default

The CAPF server is not enrolled with the CA or RA.

Command Modes

CAPF-server configuration (config-capf-server)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example specifies that the CAPF server should enroll with the trustpoint named server12 (the CA) using the password x8oWiet, which should be encrypted:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

clear cti session

To tear down the connection between a CSTA client application and Cisco Unified CME, use the **clear cti session** command in privileged EXEC configuration mode.

clear cti session *session_id*

Syntax Description

session_id	Unique numeric identifier for the session. String length is 1 to 10 characters. String value is 1 to 2147483647.

Command Default

The CTI session between the application and the router is active.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command gracefully disassociates the connection between a CSTA application and Cisco Unified CME. Use this command to direct Cisco Unified CME to send a SIP BYE for the CSTA call to the application and to clean up the session internally. This command does not reset the IP phone.

To disassociate the connection without using this command, you must reboot the router or the CSTA client application.

This command has a **no** form, but the **no** form has no effect.

To determine the ID for an active CTI session, use the **show cti session** command.

Examples

The following example shows how to tear down session 10133 between a CSTA client application and Cisco Unified CME:

```
Router# clear cti session 10133
Router#
```

Related Commands

Command	Description
show cti session	Displays active CTI sessions.

clear telephony-service conference hardware number

To drop all conference parties and clear the conference call, use the **clear telephony-service conference hardware number** command in privileged EXEC mode.

clear telephony-service conference hardware number *number*

Syntax Description

number	Conference telephone or extension number.

Command Default

The conference call continues with all current parties.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ2	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Use the **show telephony-service conference hardware** command to display the active hardware conferences. Use the **clear telephony-service conference hardware number** command to clear the desired conference.

Examples

The following example clears the conference number 1111 and drops all its parties:

```
Router# clear telephony-service conference hardware number 1111
```

Related Commands

Command	Description
show telephony-service conference hardware	Displays information about hardware conferences in a Cisco CME system.

clear telephony-service ephone-attempted-registrations

To empty the log of ephones that unsuccessfully attempt to register with Cisco Unified CME, use the **clear telephony-service ephone-attempted-registrations** command in privileged EXEC configuration mode.

clear telephony-service ephone-attempted-registrations

Syntax Description

This command has no keywords or arguments.

Command Default

The log continues to accumulate attempted ephone registrations.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **no auto-reg-ephone** command blocks the automatic registration of ephones whose MAC addresses are not explicitly listed in the configuration. When automatic registration is blocked, Cisco Unified CME records the MAC addresses of phones that attempt to register but cannot because they are blocked.

Use the **show ephone attempted-registrations** command to view the list of phones that have attempted to register but have been blocked. The **clear telephony-service ephone-attempted-registrations** command clears the list.

Examples

The following example clears the attempted-registrations log.

```
Router# clear telephony-service ephone-attempted-registrations
```

Related Commands

Command	Description
auto-reg-ephone	Enables automatic registration of ephones with Cisco Unified CME.
show ephone attempted-registrations	Displays the log of ephones that unsuccessfully attempt to register with Cisco CME.

clear telephony-service xml-event-log

To clear the event table used for the Cisco Unified CME XML application, use the **clear telephony-service xml-event-log** command in privileged EXEC mode.

clear telephony-service xml-event-log

Syntax Description

This command has no keywords or arguments.

Command Default

The XML event table is not cleared.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **show fb-its-log** command displays the contents of the XML event table.

Examples

The following example clears the entries from the XML event table:

```
Router# clear telephony-service xml-event-log
```

Related Commands

Command	Description
show fb-its-log	Displays Cisco Unified CME XML API information.

clear voice fac statistics

To clear the voice FAC statistics information, use the clear voice fac statistics command in user EXEC or privileged EXEC mode.

clear voice fac statistics

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or value.

Command Modes

Privileged EXEC.

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to clear the voice Forced Authentication Code (FAC) statistics information collected by the system.

```
Router #clear voice fac statistics
```

Related Commands

Command	Description
show voice fac statistics	Displays details of phones that attempted to register and failed.

clear voice lpcor statistics

To clear all logical partitioning class of restriction (LPCOR) statistics that are displayed when the **show voice lpcor statistics** command is used, use the **clear voice lpcor statistics** command in privileged EXEC mode.

clear voice lpcor statistics

Syntax Description

This command has no arguments or keywords.

Command Default

Statistics continue to increment.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command resets all LPCOR failed-call statistics to 0. Use the **show voice lpcor statistics** command to display the current statistics.

Examples

The following example resets the LPCOR statistics:

```
Router# clear voice lpcor statistics
```

Related Commands

Command	Description
show voice lpcor statistics	Displays information about LPCOR policies and calls.
voice lpcor policy	Creates a LPCOR policy for a resource group.

clear voice moh-group statistics

To clear the display of MOH subsystem statistics information and reset the packet counters, use the **clear voice moh-group statistics** command in privileged EXEC mode.

clear voice moh-group statistics

Syntax Description

This command has no arguments or keywords

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	yCisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use this command to clear the display of MOH subsystem statistics information displayed by the show voice moh-group statistics command.

We recommend that the clear voice moh-group statistics should be used once every two years to reset the packet counters. Each packet counter is of 32 bit size limit and the largest count a packet counter can hold is 4294967296 intervals. This means that with 20 milliseconds packet interval (for G.711), the counters will restart from 0 any time after 2.72 years (2 years and 8 months).

Examples

```
Router# clear voice moh-group statistics
All moh group stats are cleared
```

Related Commands

Command	Description
show voice moh-group statistics	Displays the MOH subsystem statistics information
show voice moh-group	Displays the MOH groups configured

clear voice register attempted-registrations

To clear the attempted-registrations, use the clear voice register attempted-registrations command in voice register global mode.

clear voice register attempted registrations [{**ip** **ip-address** | **mac** **H.H.H**}]

Syntax Description

ip ip-address	(Optional) IP address of the SIP phone attempting to register.
mac H.H.H	(Optional) MAC address of the SIP phone attempting to register.

Command Default

The attempted-registration entries are not cleared.

Command Modes

Privileged EXEC.

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use this command to delete the entries in the attempted-registration table. The clear voice register attempted-registrations command does not alter the table size, but clears the existing entries. A user confirmation is sought before the cleanup is done.

The primary key to recognize the SIP phones that fail to register is through their MAC address (hardware address) and the secondary key is the IP address. You can clear the attempted registration entry for a specific phone that failed to register by providing its IP address or MAC address and create more space for new attempted registration entries in the attempted-registrations table. When no options (IP or MAC) are selected, all the entries are removed. A user confirmation is sought in such a case, before clearing the attempted-registrations table.

The ip keyword allows you to delete entries corresponding to a specific IP address. Similarly, the mac keyword allows you to clear the entries related to a specific MAC address. User confirmation is not sought if ip or mac option is used.

Examples

```
Router # clear voice regis attempted-registrations
This will clear all the entries. Proceed? Yes/No? [no]: Yes
```

```
Router# clear voice register attempted-registrations ?
  ip   IP Address of the phone
  mac  MAC Address of the phone
```

Related Commands

Command	Description
attempted-registrations size	Allows to set the size of the attempted-registrations table.
show voice register attempted-registration	Displays details of phones that attempted to register and failed.

cnf-file

To specify the generation of different phone configuration files by type of phone or by individual phone, use the **cnf-file** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

cnf-file {**perphonetype** | **perphone**}
no cnf-file {**perphonetype** | **perphone**}

Syntax Description

perphonetype	A separate configuration file is generated for each type of phone.
perphone	A separate configuration file is generated for each phone.

Command Default

A single configuration file is used for all phones.

Command Modes

Telephony-service (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Configuration files can be applied in the following ways:

- Per system—All phones use a single configuration file. This is the default behavior for Cisco Unified CME and does not require you to configure this command. The default user and network locale in a single configuration file are applied to all phones in the Cisco Unified CME system. Alternative and user-defined user and network locales are not supported. To use the per-system option after configuring this command, use the **no cnf-file** command to reset the option to default behavior.
- Per phone type—Creates separate configuration files for each phone type. For example, all Cisco Unified IP Phone 7960s use XMLDefault7960.cnf.xml, and all Cisco Unified IP Phone 7905s use XMLDefault7905.cnf.xml. All phones of the same type use the same configuration file which is generated using the default user and network locale. This option is not supported if the **cnf-file location** is configured for system.
- Per phone—Creates a separate configuration file for each phone, by MAC address. For example, a Cisco Unified IP Phone 7960 with the MAC address 123.456.789 creates the per-phone configuration file SEP123456789.cnf.xml. The configuration file for a phone is generated with the default user and network locale unless a different user and network locale is applied to the phone using an ephone template. This option is not supported if the location option is system.

To reset the type of configuration file to the default, use the **no** form of this command and the keyword that you previously used to set the type.

This feature is supported only on the following phones:

- Cisco Unified IP Phones 7940 and 7940G
- Cisco Unified IP Phones 7960 and 7960G
- Cisco Unified IP Phone 7970G
- Cisco Unified IP Phone 7971G-GE

Examples

The following example selects flash memory as the configuration file storage location and per-phone as the type of configuration files that the system should generate.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
```

Command	Description
cnf-file location	Specifies a storage location for XML configuration files.
create cnf	Generates the XML configuration files used for provisioning SCCP phones.

cnf-file location

To specify a storage location for phone configuration files, use the **cnf-file location** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

cnf-file location {**flash: | slot0: | tftp***tftp-url*}
no cnf-file location {**flash: | slot0: | tftp***tftp-url*}

Syntax Description

flash:	Router flash memory.
slot0:	Router slot 0 memory.
tftp *tftp-url*	External TFTP server at the specified URL.

Command Default

A single phone configuration file is stored in system memory and is used by all phones.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

TFTP does not support file deletion. When configuration files are updated, they overwrite any existing configuration files with the same name. If you change the configuration file location, files are not deleted from the TFTP server.

You can specify any of the following locations in which to store configuration files:

- System—This is the default. When the system is the storage location, there is only one default configuration file and it is used for all phones in the system. All phones, therefore, use the same user locale and network locale. User-defined user and network locales are not supported. To use the system location, do not use this command to specify a locatio other than system or use the no version of this command to reset the option from a previous, different location.

If VRF Support on Cisco Unified CME is configured and the **cnf-file location** command is configured for **system:**, the configuration file for an ephone in a VRF group is created in *system:/its/vrf<group-tag>/* . The vrf group directory is created and appended to the TFTP path automatically. No action is required on your part. The location for locale files is not affected. Locale files are created in system:/its/.

- Flash or slot 0—When flash or slot 0 memory on the router is the storage location, you can create additional configuration files that can be applied per phone type or per individual phone. Up to five user-defined user and network locales can be used in these configuration files. To store configuration files in flash or slot 0, use the **cnf-file location flash:** or **cnf-file location slot0:** command. The generation of configuration files on flash or slot 0 can take up to a minute, depending on the number of files that are being generated.

If VRF Support on Cisco Unified CME is configured and the **cnf-file location** command is configured as **flash:** or **slot0:**, the per phone or per phone type file for an ephone in a VRF group is named *flash:/its/vrf<group-tag>_<filename>* or *slot0:/its/vrf<group-tag>_filename>* . The vrf group directory is created and appended to the TFTP path automatically. No action is required on your part. The location for locale files is not affected. Locale files are created in flash:/its/ or in slot0:/its

Note When the storage location chosen is flash and the file system type on this device is Class B(LEFS), make sure to check free space on the device periodically and use the **squeeze** command to free the space used up by deleted files. Unless you use the **squeeze** command, the space used by the moved or deleted configuration files cannot be used by other files.

- TFTP—When an external TFTP server is the storage location, you can create additional configuration files that can be applied per phone type or per individual phone. Up to five user-defined user and network locales can be used in these configuration files. To store configuration files on an external TFTP server, use the **cnf-file location tftp** *url* command.

Examples

The following example selects flash memory as the configuration file storage location and per-phone as the type of configuration files that the system should generate.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
```

Related Commands

Command	Description
cnf-file	Specifies the use of different phone configuration files by type of phone or by individual phone.
create cnf	Generates the XML configuration files used for provisioning SCCP phones.

codec (ephone)

To select a preferred codec for Cisco Unified CME to use when configuring calls for a phone, use the **codec** command in ephone or ephone-template configuration mode. To return to the command default, use the **no** form of this command.

codec {**g711ulaw** | **g722r64** | **g729r8** [**dspfarm-assist**] | **ilbc**}
no codec

Syntax Description

g711ulaw	Preferred codec: G.711 micro-law 64K bps.
g722r64	Preferred codec: G.722-64K bps.
g729r8	Preferred codec: G.729-8K bps.
dspfarm-assist	(Optional) DSP-farm resources are used for transcoding the segment between the phone and the Cisco Unified CME router if G.711 is negotiated for the call. **Note** The **dspfarm-assist** keyword is ignored if the SCCP endpoint type is ATA, VG224, or VG248.
ilbc	Preferred codec: iLBC 20ms.

Command Default

G.711 micro-law is the preferred codec.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(15)XZ	Cisco Unified CME 4.3	The **g722r64** and **ilbc** keywords were added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command can be used to help save network bandwidth for a remote IP phone.

For calls to phones that are not in the same Cisco Unified CME system (such as VoIP calls), the codec is negotiated based on the protocol that is used for the call (such as H.323). The Cisco Unified CME system plays no part in the negotiation.

The G.722-64K codec is supported on some varieties of phone models. Check your phone documentation to make sure the phone supports the G.722-64K codecs.

The telephone's firmware version must support the specified codec. If a codec is specified by using this command and a phone does not support the preferred codec, then the phone will use the global codec as specified by using the **codec** command in telephony-service configuration mode or if the global codec is not supported, G.711 micro-law.

For calls to other phones in the same Cisco Unified CME system, an IP phone that is configured to use G.729 will always have its calls set up to use G.729. If the phone participates in a call on a line that is shared with a phone that is configured for G.729 or is paged together with another phone that is configured for G.729, it must use G.729.

When you use the **codec** command without the **dspfarm-assist** keyword, you affect only calls between two phones on the Cisco Unified CME router (such as between an IP phone and another IP phone or between an IP phone and a FXS analog phone). The command has no effect on a call directed through a VoIP dial peer unless you use the **dspfarm-assist** keyword.

When you use the **g729r8** keyword to select the G.729r8 codec for the RTP segment between the IP phone and the Cisco Unified CME router and you also use the **dspfarm-assist** keyword, the router attempts to use DSP-farm resources in the following way: If the IP phone is in a VoIP call (H.323 or SIP) or a Cisco Unified CME conference in which the codec must be set to G.711, the router uses configured DSP-farm resources to attempt to return the segment between the phone and the Cisco Unified CME router to G.729. Adequate DSP resources must be appropriately configured separately.

f the **dspfarm-assist** keyword is configured for a phone and a DSP resource is not available when needed for transcoding, a phone registered to the local Cisco Unified CME router will use G.711 instead of G.729r8. This is not true for non-SCCP call legs; if no DSP resource is available for the transcoding required for a conference, for example, the conference will not be created.

It is recommended that the **dspfarm-assist** keyword be used sparingly and only when absolutely required for bandwidth savings or when you know the phone will have few calls that require a G.711 codec.

You should consider your options carefully when deciding to use the **dspfarm-assist** keyword with the **codec** command. The benefit is that it allows calls to use the G.729r8 codec on the call leg between the IP phone and the Cisco Unified CME router, which saves network bandwidth. The disadvantage is that for situations requiring G.711 codecs, such as conferencing and Cisco Unity Express, DSP resources that can be scarce will be used to transcode the call, and delay will be introduced while voice is shuttled to and from the DSP. In addition, the overuse of this feature can mask configuration errors in the codec selection mechanisms involving dial peers and codec lists.

For information about configuring DSP-farm resources, see the Cisco Unified CME Administrator Guide .

Note The **dspfarm-assist** keyword is ignored if the SCCP endpoint type is ATA, VG224, or VG248.

This command can also be configured in ephone-template configuration mode. If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example selects the G.729 codec with DSP farm assist for calls that are being configured for ephone 25:

```
ephone 25
 button 1:37
 codec g729r8 dspfarm-assist
```

The following example uses ephone template 1 to apply the G.729 codec preference to ephone 25:

```
ephone-template 1
 codec g729r8
```

```
ephone 25
 button 1:37
 ephone-template 1
```

Related Commands

Command	Description
dspfarm (dspfarm)	Enables digital-signal-processor (DSP) farm service
dsp services dspfarm	Specifies the NM-HDV or NM-HDV-FARM on which DSP-farm services are to be enabled.
dspfarm transcoder maximum sessions	Specifies the maximum number of transcoding sessions to be supported by a DSP farm.
show dspfarm	Displays summary information about DSP resources.

codec (telephony-service)

To select a default codec for SCCP IP phones in Cisco Unified CME, use the **codec** command in telephony-service configuration mode. To disable the codec, use the **no** form of this command.

codec {**g711ulaw** | **g722r64**}
no codec

Syntax Description

g711-ulaw	Preferred codec: G.711 micro-law.
g722-64k	Preferred codec: G.722 64K bps.

Command Default

The default is G.711 micro-law.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command can be used to help save network bandwidth for a remote IP phone.

The G.722-64K codec is supported on certain phones only, such as the Cisco Unified IP Phone 7906G, 7911G, 7941G-GE, 7942G, 7945G, 7961G-GE, 7962G, 7965G, and 7975G. Check your phone documentation to make sure your phones support the G.722-64K codec.

The telephone firmware version on a Cisco Unified IP phone must support the specified codec. If this command is configured and a phone does not support the specified codec, the default codec for that phone is G.711 micro-law.

Examples

The following example shows how to configure a G.722-64K codec in telephony-service configuration mode:

```
Router(config)# telephony-service
Router(config-telephony)# codec g722r64
Router(config-telephony)# service phone g722CodecSupport 2
Router(config-telephony)#
```

Related Commands

Command	Description
service phone	Modifies VendorConfig parameters in configuration files for IP phones.

conference (ephone-dn)

To configure a conference associated with a directory number, use the **conference** command in ephone-dn configuration mode. To disable a conference associated with a directory number, use the **no** form of this command.

conference {**ad-hoc** [**video**] | [**meetme** [**video**] [**homogeneous**]] | **unlocked**}
no conference {**ad-hoc** [**video**] | [**meetme** [**video**] [**homogenous**]] **unlocked**}

Syntax Description

ad-hoc	Configures ad hoc conferences.
video	(Optional) Configures video conferences.
meetme	Configures meet-me conferences.
homogenous	(Optional) Enables a homogeneous video conference in which all participants use the same video format. **Note** The video keyword must be specified in the command.
unlocked	Unlocks the meet-me conference bridge.

Command Default

No conference is associated with the directory number.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ2	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. The command output was enhanced to display the unlocked meet-me conference setting.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(4)M	Cisco Unified CME 8.6	This command was modified to configure video conferences.
Cisco IOS XE Everest 16.5.1b	Unified CME 11.7	This command was integrated into Cisco IOS XE Everest 16.5.1 Release to support Cisco 4000 Series Integrated Services Router.

Usage Guidelines

Ad hoc conferences are those that begin as a call between the conference creator and another party. The creator then calls other parties and adds them to the original call creating a conference.

Meet-me conferences have a designated meet-me telephone or extension number that all parties call to join the conference. The conference creator initiates the meet-me conference by pressing the MeetMe softkey, then dialing the meet-me number. Other parties join the conference by dialing the meet-me number. Homogenous video conferences only applies to meet-me conferences.

An unlocked meet-me conference allows the user to unlock the meet-me conference bridge. All DN tags with the same number should be configured with the unlocked option. Unlocking the meet-me conference bridge can allow unrestricted and uncontrolled access for the external callers. This feature is support only for meet-me conferences.

When you unlock meet-me conference bridge in Cisco Unified CME, the user can initiate a meet-me conference without pressing the MeetMe softkey, which would allow the external callers to initiate a meet-me conference.

Note To configure an unlocked meet-me conference, all ephone-dn tags associated with the same number should have the unlocked option configured. If some of the ephone-dn tags do not have the unlocked option configured, the unlocked meet-me conference may not work properly.

Use the **ephone-dn** command to configure enough extensions for your conference needs. Each extension can handle two conference parties if the **dual-line** keyword is used with the **ephone-dn** command, as shown in the following example. Use the **show ephone-dn** command to display phone information for the extension.

Examples

The following example configures extension 9001 as a four-party meet-me conference number.

```
Router(config)# ephone-dn 1 dual-line
Router(config-ephone-dn)# number 9001
Router(config-ephone-dn)# conference meetme
Router(config-ephone-dn)# no huntstop
Router(config)# ephone-dn 2 dual-line
Router(config-ephone-dn)# number 9001
Router(config-ephone-dn)# conference meetme
Router(config-ephone-dn)# preference 1
```

You must configure additional directory numbers to add more parties to the conference.

Related Commands

Command	Description
show ephone-dn	Displays phone information for specified dn-tag or for all dn-tags.

conference (voice register template)

To enable the soft key for conference in a SIP phone template, use the **conference** command in voice register template configuration mode. To disable the soft key, use the **no** form of this command.

conference
no conference

Syntax Description

This command has no arguments or keywords.

Command Default

Soft key for conference is enabled.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command enables a soft key for conference in the specified template which can then be applied to SIP phones. The conference soft key is enabled by default. To disable the conference soft key, use the **no conference** command. To apply a template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following example shows how to disable the conference soft key in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# no conference
```

Related Commands

Command	Description
template (voice register pool)	Applies a template to a SIP phone.
transfer-attended (voice register template)	Enables a soft key for attended transfer in a SIP phone template.
transfer-blind (voice register template)	Enables a soft key for blind transfer in a SIP phone template.
voice register pool	Enters voice register pool configuration mode for SIP phones.

conference add-mode

To configure the mode for adding parties to ad hoc hardware conferences, use the **conference add-mode** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

conference add-mode [**creator**]
no conference add-mode [**creator**]

Syntax Description

creator	Specifies that only the creator can add parties.

Command Default

Any party can add other parties provided the creator remains in the conference.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ2	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
Cisco IOS XE Everest 16.5.1b	Unified CME 11.7	Support for this command was introduced on the Cisco 4000 Series Integrated Services Routers.

Usage Guidelines

For more control of conference participation, use this command to specify that only the creator can add new parties. This configuration ensures that no one can add parties to the conference without the creator’s knowledge.

Use this command to configure an ephone directly in ephone configuration mode, or use it to configure an ephone template in ephone-template configuration mode. Use the **ephone-template** command in ephone configuration mode to apply the ephone template to one or more ephones. Use the **show telephony-service ephone** command to display the add and drop modes for the ephone. Use the **show telephony-service ephone-template** command to display the ephone template.

Examples

The following example configures ad hoc hardware conferences so that only the creator can add participants.

```
Router(config)# ephone 1
Router(config-ephone)#  conference add-mode creator
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an ephone.
show telephony-service ephone	Displays configuration for the Cisco IP phones.
show telephony-service ephone-template	Displays the contents of ephone-templates.

conference add-mode (voice register)

To configure the mode for adding participants to ad-hoc hardware conferences on Cisco Unified SIP IP phones, use the **conference add-mode** command in voice register pool or voice register template configuration mode. To return to the default, use the **no** form of this command.

conference add-mode [**creator**]
no conference add-mode

Syntax Description

creator	(Optional) Specifies that only the conference creator can add participants to an ad-hoc hardware conference.

Command Default

The conference creator or any of the participants can add a new participant.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

Command History

Release	Modification
15.2(2)T	This command was introduced.
Cisco IOS XE Everest 16.5.1b	Support for this command was introduced on the Cisco 4000 Series Integrated Services Routers.

Usage Guidelines

Use the **conference add-mode creator** command to specify that only the conference creator can add new participants. This configuration ensures that no one can add participants to the hardware conference without the creator's knowledge.

Examples

The following example shows how to configure the mode so that only the conference creator can add new participants to a hardware conference on voice register pool 10:

```
Router(config)# voice register pool 10
Router(config-register-pool)# conference add-mode creator
```

The following example shows how to configure the mode so that only the conference creator can add new participants to a hardware conference on template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# conference add-mode creator
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.

conference admin

To configure the ephone as the ad hoc and meet-me hardware conference administrator, use the **conference admin** command in ephone or ephone-template configuration mode. To return to the defaults, use the **no** form of this command.

conference admin
no conference admin

Syntax Description

This command has no arguments or keywords.

Command Default

This ephone is not the ad hoc and meet-me hardware conference administrator.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ2	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T

Usage Guidelines

Use this command to configure an ad hoc and meet-me hardware conference administrator. The administrator can:

- Dial in to any conference directly through the conference number
- Use the ConfList soft key to list conference parties
- Remove any party from any conference

The administrator can control the use of conference bridges by enforcing time limits and making sure conference bridges are available for scheduled meetings.

Use this command to configure an ephone directly in ephone configuration mode, or use it to configure an ephone template in ephone-template configuration mode. Use the **ephone-template** command in ephone configuration mode to apply the ephone template to one or more ephones. Use the **show telephony-service ephone** command to display the add and drop modes for the ephone. Use the **show telephony-service ephone-template** command to display the ephone template.

Examples

The following example configures ephone 1 as the ad hoc and meet-me hardware conference administrator.

```
Router(config)# ephone 1
Router(config-ephone)#  conference admin
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an ephone.
show telephony-service ephone	Displays configuration for the Cisco IP phones.

Command	Description
show telephony-service ephone-template	Displays the contents of ephone-templates.

conference admin (voice register)

To assign a Cisco Unified SIP IP phone as the ad-hoc or meet-me hardware conference administrator, use the **conference admin** command in voice register pool or voice register template configuration mode. To return to the default, use the **no** form of this command.

conference admin
no conference admin

Syntax Description

This command has no arguments or keywords.

Command Default

The Cisco Unified SIP IP phone is not the conference administrator.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **conference admin** command to assign an ad-hoc or meet-me hardware conference administrator. The administrator can:

- Dial in to any conference directly through the conference number.
- Use the ConfList soft key to list conference participants.
- Remove any participant from any conference.

The administrator can control the use of conference bridges by enforcing time limits and making sure conference bridges are available for scheduled meetings.

Examples

The following example shows how to configure voice register pool 25 as the conference administrator:

```
Router(config)# voice register pool 25
Router(config-register-pool)# conference admin
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.

conference drop-mode

To configure the mode for terminating ad hoc hardware conferences when parties drop out, use the **conference drop-mode** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

conference drop-mode [{**creator** | **local**}]
no conference drop-mode [{**creator** | **local**}]

Syntax Description

creator	Specifies that the active conference terminates when the creator hangs up.
local	Specifies that the active conference terminates when the last local party in the conference hangs up or drops out of the conference.

Command Default

The conference is not dropped, regardless of whether the creator hangs up, provided three parties remain in the conference.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ2	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T
Cisco IOS XE Everest 16.5.1b	Unified CME 11.7	Support for this command was introduced on the Cisco 4000 Series Integrated Services Routers.

Usage Guidelines

For more control of conference participation, use this command to specify that the conference drops when the creator hangs up (see the example). This configuration ensures that the conference cannot continue without the creator's presence.

Use this command to configure an ephone directly in ephone configuration mode, or use it to configure an ephone template in ephone-template configuration mode. Use the **ephone-template** command in ephone configuration mode to apply the ephone template to one or more ephones. Use the **show telephony-service ephone** command to display the add and drop modes for the ephone. Use the **show telephony-service ephone-template** command to display the ephone template.

Examples

The following example configures ad hoc hardware conferences so that only the creator can add participants and the active conference terminates when the creator hangs up.

```
Router(config)# ephone 1
Router(config-ephone)#  conference drop-mode creator
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an ephone.
show telephony-service ephone	Displays configuration for the Cisco IP phones.
show telephony-service ephone-template	Displays the contents of ephone-templates.

conference drop-mode (voice register)

To specify who can terminate an active ad-hoc hardware conference by hanging up, use the **conference drop-mode** command in voice register pool or voice register template configuration mode. To return to the default, use the **no** form of this command.

conference drop-mode {**creator** | **local**}
no conference drop-mode

Syntax Description

creator	Terminates the active conference when the conference creator hangs up.
local	Terminates the active conference when the last local participant hangs up or drops out of the conference.

Command Default

An active conference is never dropped.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

Command History

Release	Modification
15.2(2)T	This command was introduced.
Cisco IOS XE Everest 16.5.1b	Support for this command was introduced on the Cisco 4000 Series Integrated Services Routers.

Usage Guidelines

Use the **conference drop-mode creator** command to specify that an active hardware conference is terminated when the creator hangs up. This configuration ensures that the hardware conference cannot continue without the creator's presence.

Examples

The following example shows how to configure an active conference so that it is terminated when the conference creator hangs up:

```
Router(config)# voice register pool 60
Router(config-register-pool)# conference drop-mode creator
```

The following example shows how to configure an active conference so that it is terminated when the last local participant hangs up or drops out of the conference:

```
Router(config)# voice register template 7
Router(config-register-temp)# conference drop-mode local
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.

Command	Description
voice register template	Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.

conference hardware

To configure a Cisco Unified CallManager Express system for hardware conferencing only, use the **conference hardware** command in telephony-service configuration mode. To return to the default three-party software conferencing, use the **no** form of this command.

conference hardware
no conference hardware

Syntax Description

This command has no arguments or keywords.

Command Default

Three-party ad hoc software conferencing.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ2	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
Cisco IOS XE Everest 16.5.1b	Unified CME 11.7	This command was integrated into Cisco IOS XE Everest 16.5.1 Release to support Cisco 4000 Series Integrated Services Router.

Usage Guidelines

Software conferencing allows a maximum of three parties in a conference. Use this command to take advantage of DSP farm resources for hardware conferencing that allows ad hoc conferences with more than three parties.

If you need ad hoc hardware conferences, you must use this command to configure DSP farm hardware conferencing. You can configure other conferencing features using the **conference-join custom-cptone**, **conference-leave custom-cptone**, and **maximum conference-participants** commands in DSP farm profile configuration mode. Use the **show dspfarm profile** command to display the DSP farm profile.

Examples

The following example configures hardware conferencing as the default for ad hoc conferences on this Cisco Unified CallManager Express system:

```
Router(config)# telephony-service
Router(config-telephony)# conference hardware
```

Related Commands

Command	Description
conference-join custom-cptone	Associates a custom call-progress tone to indicate joining a conference with a DSP farm profile.
conference-leave custom-cptone	Associates a custom call-progress tone to indicate leaving a conference with a DSP farm profile.

Command	Description
maximum conference-participants	Configures the maximum number of conference participants allowed in each conference.
show dspfarm profile	Displays configured DSP farm profile information.

conference hardware (voice register global)

To configure Cisco Unified Communications Manager Express (Cisco Unified CME) DSPFarm hardware-based ad-hoc conferencing, use the **conference hardware** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

conference hardware
no conference hardware

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco Unified SIP IP phone local conference is enabled.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
15.2(2)T	Cisco Unified CME 9.0	This command was introduced.
Cisco IOS XE Everest 16.5.1b	Unified CME 11.7	This command was integrated into Cisco IOS XE Everest 16.5.1 Release yto support Cisco 4000 Series Integrated Services Router.

Usage Guidelines

Use the **conference hardware** command in voice register global configuration mode to take advantage of DSPfarm resources that allow ad-hoc hardware conferences with more than three parties.

Enable hardware conferencing in telephony-service configuration mode before configuring hardware conferencing in voice register global configuration mode. Otherwise, the configuration of hardware conferencing in voice register global configuration mode will be rejected.

If you apply any changes to the configuration of the hardware conference, you must use the **create profile** command in voice global configuration mode to regenerate the configuration profile files required for Cisco Unified SIP IP phones. Then, reboot the phone.

Examples

The following example shows how to configure Cisco Unified CME DSPFarm hardware-based ad-hoc conferencing:

```
Router(config)# telephony-service
Router(config-telephony)# conference hardware
.
Router(config)# voice register global
Router(config-register-global)# conference hardware
```

Related Commands

Command	Description
conference hardware	Configures a Cisco Unified CME system for hardware conferencing only in telephony-service configuration mode.

conference max-length

To allow conferencing, only if the number of dialed digits are within the max-length limit, use the **conference max-length** command. To remove the configuration, use the **no** form of this command.

conference max-length <*value*>
no conference max-length

Syntax Description

value	Maximum number of digits that can be dialed. The range is from 3 to 16.

Command Default

By default, no value is defined for conferencing.

Command Modes

Ephone (config-ephone)

Ephone-template ephone (config-ephone-template)

Voice register pool (config-register-pool)

Voice register template(config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

Use the **conference max-length** command to configure, the Cisco Unified CME to allow conferencing, only if the dialed digits are within the maximum length limit.

Example

The following example shows how to configure the maximum length of 8 digits that can be dialed to make a conference call:

```
Router(config)# ephone 1
Router(config-ephone)# conference max-length 8
```

Related Commands

Command	Description
conference-pattern blocked	Blocks extensions on an ephone or a voice register pool from making conference calls.
conference transfer-pattern	Apply transfer -pattern configuration for conference cases.
transfer max-length	Allows transfer of calls to phones, where the number of dialed digits are less than the maximum length configured.
transfer-pattern (telephony-service)	Allows the transfer of calls to phones outside the Cisco Unified CME network.

conference-pattern blocked

To prevent extensions on an ephone or a voice register pool from initiating a conference to external numbers, use the **conference-pattern blocked** command. Note that the **conference-pattern blocked** command does not impact call transfer functions. To remove the configuration, use the **no** form of this command.

conference-pattern blocked
no conference-pattern blocked

Syntax Description

This command has no arguments or keywords.

Command Default

No default values are defined.

Command Modes

Ephone configuration (config-ephone)

Ephone-template configuration (config-ephone-template)

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

Use the **conference-pattern blocked**command to prevent specific extensions from making conference calls to patterns generally allowed through the **transfer-pattern** command.

Example

The following example shows how to prevent extensions from making conference calls using the **conference-pattern blocked**command:

```
Router(config)# ephone 1
Router(config-ephone-template)# conference-pattern blocked
```

Related Commands

Command	Description
conference max-length	Allows conferences to numbers where dialed digits are within the configured maximum length value.
conference transfer-pattern	Apply transfer-pattern configuration for conference cases.
transfer-pattern blocked	Blocks individual phones from transferring calls to nonlocal numbers that have been globally enabled for transfer.
transfer-pattern (telephony-service)	Allows the transfer of calls to phones outside the Cisco Unified CME network.

conference transfer-pattern

To configure a Cisco Unified CallManager Express system to apply transfer-patter <pattern> to the conference call using conference softkey or feature button, use the **conference transfer-pattern** command in telephony-service configuration mode. To return to the default, use the no form of this command.

conference transfer-pattern
no conference transfer-pattern

Syntax Description

This command has no arguments or keywords.

Command Default

Transfer-pattern <pattern> does not apply to call conferencing.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
15.2(4)M	Cisco Unified CME 9.1	This command was introduced.

Usage Guidelines

There is no check for the conference numbers for call conferencing. Use this command to apply transfer-pattern for call conferencing.

Examples

The following example enables transfer-pattern to be applied for conference parties:

```
Router(config)# telephony-service
Router(config-telephony)# conference transfer-pattern
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

cor (ephone-dn)

This command is now documented as the **corlist** command. For complete command information, see the **corlist** command page.

cor (voice register)

To configure a class of restriction (COR) on the VoIP dial peers associated with directory numbers, use the **cor** command in voice register pool or voice register template configuration mode. To disable a COR associated with directory numbers, use the **no** form of this command.

cor **{incoming | outgoing}***cor-list-name {cor-list-number starting-number [- ending-number]*|**default***}*
no cor **{incoming | outgoing}***cor-list-name {cor-list-number starting-number [- ending-number]*||**default***}*

Syntax Description

incoming	COR list to be used by incoming dial peers.
outgoing	COR list to be used by outgoing dial peers.
cor-list-name	COR list name.
cor-list-number	COR list identifier.
starting-number	Start of a directory number range, if an ending number is included. Can also be a standalone number.
-	(Optional) Indicator that a full range is configured.
ending-number	(Optional) End of a directory number range.
default	Instructs the COR list to assume behavior according to a predefined default COR list.

Command Default

COR is not configured on VoIP dial peers.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CallManager Express (Cisco CME).
Cisco IOS XE Fuji 16.7.1	Unified CME 12.1	This command was supported in voice register template configuration mode.

Usage Guidelines

The **cor** command sets the dial-peer COR parameter for dynamically created VoIP dial peers. A list-based mechanism assigns COR parameters to specific set of number ranges. The COR functionality provides the ability to deny certain call attempts on the basis of the incoming and outgoing class of restrictions provisioned on the dial peers. This functionality provides flexibility in network design, allows users to block calls (for example, calls to 900 numbers), and applies different restrictions to call attempts from different originators.

COR specifies which incoming dial peer can use which outgoing dial peer to make a call. Each dial peer can be provisioned with an incoming and an outgoing COR list.

A default COR is assigned to the directory numbers that do not match any COR list number or number range. During Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) registration, a dial peer is created and that dial peer includes a default COR value. The **cor** command allows you to change the automatically selected default.

In dial-peer configuration mode, build your COR list and add members. Then in voice register pool configuration mode, use the **cor** command to apply the name of the dial-peer COR list.

If the **cor** command is configured under voice register template and voice register pool configuration modes, precedence is for the COR configuration under voice register pool configuration mode.

You can have up to four COR lists for the Cisco Unified SIP SRST configuration, comprised of incoming or outgoing dial peers. The first four COR lists are applied to a range of phone numbers. The phone numbers that do not have a COR list configuration are assigned to the default COR list, providing that a default COR list has been defined.

Note Configure the **id** (voice register pool) command before any other voice register pool commands, including the **cor** command. The **id** command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

Examples

The following is sample output from the **show running-config** command:

```
..
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
 proxy 10.2.161.187 preference 1 monitor probe icmp-ping
 alias 1 94... to 91011 preference 8
 voice-class codec 1
.
.
.
dial-peer cor custom
 name 95
 name 94
 name 91
!
dial-peer cor list call91
 member 91
!
dial-peer voice 91500 pots
 corlist incoming call91
 corlist outgoing call91
 destination-pattern 91500
 port 1/0/0
.
.
.
```

The following is a sample output of the **show running-config** for COR configured under voice register template configuration mode.

```
..
voice register template 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
 proxy 10.2.161.187 preference 1 monitor probe icmp-ping
 alias 1 94... to 91011 preference 8
 voice-class codec 1
.
.
.
dial-peer cor custom
 name 95
 name 94
 name 91
!
dial-peer cor list call91
 member 91
!
dial-peer voice 91500 pots
 corlist incoming call91
 corlist outgoing call91
 destination-pattern 91500
 port 1/0/0
.
.
```

Related Commands

Command	Description
dial-peer cor custom	Specifies that named CORs apply to dial peers.
dial-peer cor list	Defines a COR list name.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
member (dial-peer cor list)	Adds a member to a dial-peer COR list.
name (dial-peer custom cor)	Provides a name for a custom COR.
show dial-peer voice	Displays information for voice dial peers.

corlist

This command was previously documented as the **cor** command.

To apply a class of restriction (COR) to the dial peers associated with a Cisco CME extension (ephone-dn), use the **corlist command in** ephone-dn configuration mode. To disable the COR associated with an extension, use the **no** form of this command.

corlist {**incoming** | **outgoing**} *corlist-name*
no corlist {**incoming** | **outgoing**}

Syntax Description

incoming	Specifies a COR list to be used by incoming dial peers.
outgoing	Specifies a COR list to be used by outgoing dial peers.
corlist-name	COR list name.

Command Default

No COR is used by the dial peers associated with the extension that is being configured.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

COR is used to specify which incoming ephone-dn dial peer can use which outgoing ephone-dn dial peer to make a call. COR denies certain call attempts on the basis of the incoming and outgoing class of restrictions that have been provisioned on the dial peers. This functionality provides flexibility in network design, allows administrators to block calls (for example, calls to 900 numbers), and applies different restrictions to call attempts from different originators.

Each dial peer can be provisioned with an incoming and an outgoing COR list.

The **corlist incoming** and **corlist outgoing** commands in dial-peer configuration mode perform these functions for dial peers that are not associated with ephone-dns. The **dial-peer cor list** and **member** commands define the sets of capabilities, or COR lists, that are referred to in the **corlist** commands.

Examples

The following example shows how to set a COR parameter for incoming calls to dial peers associated with the extension that has the dn-tag 1:

```
Router(config)# ephone-dn 1
```

```
Router(config-ephone-dn)# corlist incoming
 corlist1
```

Related Commands

Command	Description
corlist incoming	Specifies the COR list to be used when a specified dial peer acts as the incoming dial peer.
corlist outgoing	Specifies the COR list to be used by an outgoing dial peer.
dial-peer cor list	Defines a COR list name.

create cnf-files

To build the eXtensible Markup Language (XML) configuration files that are required for IP phones in Cisco Unified CME, use the **create cnf-files** command in telephony-service configuration mode. To remove the configuration files and disable the automatic generation of configuration files, use the **no** form of this command.

create cnf-files
no create cnf-files

Syntax Description

This command has no arguments or keywords.

Command Default

Required XML configuration files are not built.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was modified to interact with the **cnf-file** command and the **cnf-file location** command.
12.4(9)T	Cisco Unified CME 4.0	Modifications to this command for interacting with the **cnf-file** command and the **cnf-file location** command were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to generate the XML configuration files used for provisioning SCCP phones and write the files to the location specified with the **cnf-file location** command.

Examples

The following example builds the necessary XML configuration files on the Cisco Unified CME router:

```
Router(config)# telephony-service
Router(config-telephony)# create cnf-files
```

Related Commands

Command	Description
cnf-file	Specifies the type of configuration file to be created.
cnf-file location	Specifies a storage location for phone configuration files

create cnf-files (voice-gateway)

To generate the eXtensible Markup Language (XML) configuration files that are required to autoconfigure the Cisco voice gateway, use the **create cnf-files** command in voice-gateway configuration mode. To disable the generating of configuration files, use the **no** form of this command.

create cnf-files
no create cnf-files

Syntax Description

This command has no arguments or keywords.

Command Default

Required XML configuration files are not built.

Command Modes

Voice-gateway configuration (config-voice-gateway)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.

Usage Guidelines

Cisco Unified CME writes the XML files generated by this command to the location specified with the **cnf-file location** command, or to the default location in system:/its/. The voice gateway downloads its configuration file from Cisco Unified CME when you run the autoconfiguration process on the voice gateway.

Examples

The following example shows that the gateway configuration files are generated by Cisco Unified CME:

```
voice-gateway system  1
 network-locale FR
 type VG224
 mac-address 001F.A30F.8331
 voice-port 0-23
 create cnf-files
```

Related Commands

Command	Description
cnf-file location	Specifies a storage location for phone configuration files.
reset (voice-gateway)	Performs a complete reboot of all analog phones associated with the voice gateway and registered to Cisco Unified CME.
restart (voice-gateway)	Performs a fast restart of all analog phones associated with the voice gateway and registered to Cisco Unified CME.

create profile (voice register global)

To generate the configuration profile files required for SIP phones, use the **create profile** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

create profile
no create profile

Syntax Description

This command has no arguments or keywords.

Command Default

Configuration files are not generated.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command generates the configuration files used for provisioning SIP phones and writes the files to the location specified with the **tftp-path** command.

Examples

The following example shows how to create the configuration profile:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# create profile
```

Related Commands

Command	Description
file text (voice register global)	Generates ASCII text files for SIP phones.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
source-address (voice register global)	Identifies the IP address and port through which SIP phones communicate with a Cisco CME router.
tftp-path (voice register global)	Specifies the directory to which the provisioning file for SIP phones in a Cisco CallManager Express (Cisco CME) system will be written.

credentials

To enter credentials configuration mode to configure a certificate for a Cisco Unified CME CTL provider or for Cisco Unified SRST router communication to Cisco Unified CallManager, use the **credentials** command in global configuration mode. To set all commands in credentials configuration mode to the default of nonsecure, use the **no** form of this command.

credentials
no credentials

Syntax Description

This command has no arguments or keywords.

Command Default

Credentials are not provided.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco SRST 3.3	This command was introduced for Cisco SRST.
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced for Cisco Unified CME.
12.4(9)T	Cisco Unified CME 4.0	This command for Cisco Unified CME was integrated into Cisco IOS Release 12.4(9)T.
Cisco IOS XE Fuji 16.7.1 Release	Unified SRST 12.1	This command was introduced for Unified SRST support on Cisco 4000 Series Integrated Services Router.

Usage Guidelines

This command is used to configure credentials service for Cisco Unified CME and Cisco Unified SRST.

Cisco Unified CME

This command is used with Cisco Unified CME phone authentication to configure a CTL provider on each Cisco Unified CME router on which the CTL client is not running. That is, if there is a primary and a secondary Cisco Unified CME router and the CTL client is running on the primary router, a CTL provider must be configured on the secondary router, and vice versa. If the CTL client is running on a router that is not a Cisco Unified CME router, CTL providers must be configured on all Cisco Unified CME routers.

Credentials service for Cisco Unified CME runs on default port 2444.

Cisco Unified SRST

The credential server provides certificates to any device that requests a certificate. The credentials server does not request any data from a client; thus no authentication is necessary. When the client, Cisco Unified CallManager, requests a certificate, the credentials server provides the certificate. Cisco Unified CallManager exports the certificate to the phone, and the Cisco Unified IP phone holds the SRST router certificate in its configuration file. The device certificate for secure SRST routers is placed in the configuration file of the Cisco Unified IP phone because the entry limit in the certificate trust list (CTL) of Cisco Unified CallManager is 32.

Credentials service for SRST runs on default port 2445. Cisco Unified CallManager connects to port 2445 on the secure SRST router and retrieves the secure SRST device certificate during the TLS handshake.

Activate this command on all SRST routers.

Caution For security reasons, credentials service should be deactivated on all SRST routers after provisioning to Cisco Unified CallManager is completed.

Examples

Cisco Unified CME

The following example configures a CTL provider on the Cisco Unified CME router with the IP address 172.19.245.1. CTL providers must be configured on all Cisco Unified CME routers on which the CTL client is not running.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint cmeca
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```

Cisco Unified SRST

The following example enters credentials configuration mode and sets the IP source address and the trustpoint:

```
Router(config)# credentials
Router(config-credentials)# ip source-address 10.6.21.4 port 2445
Router(config-credentials)#
trustpoint srstca
```

Related Commands

Command	Description
ctl-service admin	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
debug credentials	Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider the CTL client or between an SRST router and Cisco Unified CallManager.
ip source-address (credentials)	Enables the Cisco Unified CME or SRST router to receive messages through the specified IP address and port.
show credentials	Displays the credentials settings on a Cisco Unified CME or SRST router.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.

cti csta mode basic

To set the CTI interface in Cisco Unified CME into basic mode, use the **cti csta mode basic** command in voice-service configuration mode. To return to default, use the **no** form of this command.

cti csta mode basic
no cti csta mode basic

Syntax Description

This command has no arguments or keywords.

Command Default

CTI interface is in advanced mode.

Command Modes

Voice-service configuration (config-voi-serv)

Command History

Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1.(1)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	This command was deprecated. It is not supportd on Unified CME 12.6 and later releases.

Usage Guidelines

This command supresses all enhanced extensions/features, such as shared line and shared media, in a CTI message from Cisco Unified CME.

This command is required if the computer-based CSTA client application that is interacting with Cisco Unified CME is a Microsoft Office Communicator (MOC) client .

Examples

The following example shows a voice-service configuration with this command enabled:

```
!
voice service voip
 no cti shutdown
 cti csta mode basic
!
```

Related Commands

Command	Description
cti shutdown	Disables CTI integration.

cti message device-id suppress-conversion

To suppress the conversion or promotion of all extension numbers except the primary number in a CTI message, use the **cti message device-id suppress-conversion** command in voice-service configuration mode. To return to default, use the **no** form of this command.

cti message device-id suppress-conversion
no cti message device-id suppress-conversion

Syntax Description

This command has no arguments or keywords.

Command Default

All SCCP extension numbers are converted or promoted in CTI messages.

Command Modes

Voice-service configuration (config-voi-serv)

Command History

Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	This command was deprecated. It is not supportd on Unified CME 12.6 and later releases.

Usage Guidelines

This command specifies that only the requested (primary) extension number is converted or promoted in the outgoing CTI message when an expanded number is presented in a RequestSystemStatus from a CSTAclient application. Use this command to suppress the conversion or promotion of all secondary numbers in a CTI message.

By default, Cisco Unified CME converts or promotes all SCCP primary and secondary extension numbers when reporting events.

Examples

The following example shows the voice-service configuration with this command enabled:

```
!
voice service voip
 no cti shutdown
 cti csta mode basic
 cti message device-id suppress-conversion
```

Related Commands

Command	Description
cti shutdown	Disables CTI integration.

cti notify

To force an ephone-dn into a constant "up" state, use the **cti notify** command in ephone-dn or ephone-dn-template configuration mode. To return to default, use the **no** form of this command.

cti notify
no cti notify

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco Unified CME cannot send notifications to the ephone-dn because a CTI session cannot be established.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	This command was deprecated. It is not supportd on Unified CME 12.6 and later releases.

Usage Guidelines

This command forces an ephone-dn into a constant "up" state.

Use this command to permit a CTI session to be established with a directory number that is not associated with a physical device, allowing Cisco Unified CME to send notifications to the directory number. If a directory number is not associated to an ephone configuration that includes the button command, a static fwd is applied to the directory number and all incoming calls are forwarded to another directory number.

If you use an ephone-dn template to apply this command to a directory number and you also use this command in ephone-dn configuration mode for the same directory number, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example shows the configuration for ephone-dn 4 including this command. A CTI session can be established for this directory number (204) even though the number is not associated with an ephone configuration because this directory number is always "up."

```
!
ephone-dn  4
 number 204
 cti notify
 cti watch
!
!
ephone  1
 mac-address 001E.4A34.A35F
 type 7961
 button  1:1
```

```
!
!
!
ephone  2
 mac-address 000F.8FC7.B681
 type 7960
 button  1:2
!
!
!
ephone  3
 mac-address 0019.E7FF.1E30
 type 7961
 logout-profile 1
!
```

The following example shows how to create the same configuration for ephone-dn 4 using this command in ephone-dn template configuration mode and then applying the template to the directory number:

```
ephone-dn-template 15
 cti notify
 cti watch
ephone-dn 4
 number 204
 ephone-dn-template 15
```

Related Commands

Command	Description
ephone-dn-template (ephone-dn)	Applies an ephone-dn template to an ephone-dn.

cti watch

To allow a CSTA client application to monitor and control a directory number in Cisco Unified CME, use the **cti watch** command in ephone-dn or ephone-dn-template configuration mode. To return to default, use the **no** form of this command.

cti watch
no cti watch

Syntax Description

This command has no arguments or keywords.

Command Default

A CSTA client application cannot use the CTI interface to montitor and control an ephone-dn in Cisco Unified CME.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	This command was deprecated. It is not supportd on Unified CME 12.6 and later releases.

Usage Guidelines

This command enables a CSTA client application to monitor and control a directory number in Cisco Unified CME.

If you use an ephone-dn template to apply this command to a directory number and you also use this command in ephone-dn configuration mode for the same directory number, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example shows the configuration for ephone-dn 4 with this command configured. The CSTA application can montior and control the directory number (204).

```
!
ephone-dn  4
 number 204
 cti notify
 cti watch
```

The following example shows how to create the same configuration for ephone-dn 4 using this command in ephone-dn template configuration mode and applying the template to the directory number:

```
ephone-dn-template 15
 cti notify
```

```
 cti watch
ephone-dn 4
 number 204
 ephone-dn-template 15
```

Related Commands

Command	Description
ephone-dn-template (ephone-dn)	Applies an ephone-dn template to an ephone-dn

cti-aware

To bind a session to the CTI subsystem, use the **cti aware** command in voice session-server configuration mode. To return to default, use the **no** form of this command.

cti-aware
no cti-aware

Syntax Description

This command has no keywords or arguments.

Command Default

CTI-register heartbeat continues even after the CTI session is shutdown.

Command Modes

Voice session-server configuration (config-register-fs)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	This command was deprecated. It is not supportd on Unified CME 12.6 and later releases.

Usage Guidelines

This command causes the CSTA SIP keepalive response to stop if the CTI session between Cisco Unified CME and the CSTA client application expires or is down for any reason. By default, the CSTA SIP keepalive response continues even after the CTI session expires and the CSTA client application is unaware that the CTI session is not operational.

Examples

The following partial output shows the configuration for a session manager for a CSTA client application in which this command is configured:

```
router# show running-configuration

.
.
.
voice register session-server 1
 register-id app1
 keepalive 360
 cti-aware
```

Related Commands

Command	Description
keepalive (voice register session-server)	Duration for registration after which the registration expires unless the feature server or application reregisters before the registration expiry.
register-id	Creats an ID for explicitly identifying an external feature server or application during Register requests

ctl-client

To enter CTL-client configuration mode to set parameters for the CTL client, use the **ctl-client** command in global configuration mode. To return to the default, use the **no** form of this command.

ctl-client
no ctl-client

Syntax Description

This command has no keywords or arguments.

Command Default

No CTL-client parameters are set.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example defines server IP addresses and trustpoints for the CAPF server, the Cisco Unified CME router, and the TFTP server, as well as trustpoints for SAST1 and SAST2. It also specifies that a new CTL file should be generated.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

ctl-service admin

To specify a user name and password to authenticate the client during the CTL protocol, use the **ctl-service admin** command in credentials configuration mode. To return to the default, use the **no** form of this command.

ctl-service admin *username* **secret** {**0** | **1**} *password-string*
no ctl-service admin

Syntax Description

username	Defines the name that will be used to authenticate the client.
secret {**0** | **1**}	Defines a character string for login authentication and whether it will be encrypted when it is stored in the running configuration. • **0**—Not encrypted. • **1**—Encrypted using Message Digest 5 (MD5).
password-string	Character string for login authentication

Command Default

No user name or password is defined for authentication.

Command Modes

Credentials configuration (config-credentials)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to define a user who will be used to authenticate the CTL client with a CTL provider.

Examples

The following example creates a CTL provider on a Cisco Unified CME router that is not running the CTL client.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint ctlpv
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```

Command	Description
debug credentials	Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider and the CTL client or between an SRST router and Cisco Unified CallManager.
show credentials	Displays the credentials settings on a Cisco Unified CME or SRST router.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.

Cisco Unified CME Commands: D

date-format (telephony-service)

To set the date display format on the Cisco IP phones in a Cisco CallManager Express (Cisco CME) system, use the **dateformat** command in telephony-service configuration mode. To display the date in the default format, use the **no** form of this command.

date-format {**dd-mm-yy** | **mm-dd-yy** | **yy-dd-mm** | **yy-mm-dd**}
no date-format

Syntax Description

dd-mm-yy mm-dd-yy yy-dd-mm yy-mm-dd	Format in which dates are displayed on the IP phone: • **dd**—Two-digit day. • **mm**—Two-digit month. • **yy**—Two-digit year.

Command Default

Default is **mm-dd-yy.**

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Examples

The following example sets the date format to day, month, and year, so that December 17, 2004 is represented as 17-12-04.

```
Router(config)# telephony-service
Router(config-telephony)# date-format dd-mm-yy
```

date-format (voice register global)

To set the date display format on SIP phones directly connected in Cisco Unified CME, use the **dateformat** command in voice register global configuration mode. To display the date in the default format, use the **no** form of this command.

date-format {**dd-mm-yy** | **mm-dd-yy** | **yy-dd-mm** | **yy-mm-dd**}
no date-format

Syntax Description

d/m/y m/d/y y-d-m y/d/m y/m/d yy-m-d	Format in which dates are displayed on the SIP IP phone: • **d**—Two-digit date of the month • **m**—Two-digit month • **y**—Two-digit year • **yy**—Four-digit year

Command Default

Date is displayed as m/d/y.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Examples

The following example shows how to set the date format so that a date such as December 3, 2007 is represented as 2007-12-03. By using the default configuration, this same date appears as 12/03/07.

```
Router(config)# voice register global
Router(config-register-global)# date-format yy-m-d
```

Related Commands

Command	Description
dst auto-adjust (voice register global)	Enables automatic adjustment of daylight saving time on SIP phones.
time-format (voice register global)	Selects a 12-hour clock or a 24-hour clock for the time display format on SIP IP phones in Cisco Unified CME.

debug callmonitor

To collect and display debugging traces for call monitor, use the **debug callmonitor** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug callmonitor {**all** | **core** | **detail** | **errors** | **events** | **hwconf** | **info** | **xml**}
no debug command {**all** | **core** | **detail** | **errors** | **events** | **hwconf** | **info** | **xml**}

Syntax Description

all	All call-monitor debugging traces.
core	Core information debugging traces.
detail	Detailed debugging traces.
errors	Call-monitor error debugging traces.
events	Call-monitor event debugging traces.
hwconf	Debugging traces related to hardware configuration.
info	Call-monitor information debugging traces.
xml	Call-monitor XML encoding debugging traces.

Command Default

There is no default for this command.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.4(11)XW2	This command was introduced.

Examples

The following example is partial output from this command:

```
Router# debug callmonitor all
Syslog logging: enabled (11 messages dropped, 2 messages rate-limited,
                0 flushes, 0 overruns, xml disabled, filtering disabled)
No Active Message Discriminator.
No Inactive Message Discriminator.
    Console logging: disabled
    Monitor logging: level debugging, 0 messages logged, xml disabled,
                     filtering disabled
    Buffer logging:  level debugging, 444378 messages logged, xml disabled,
                     filtering disabled
    Logging Exception size (4096 bytes)
    Count and timestamp logging messages: disabled
    Persistent logging: disabled
    Trap logging: level informational, 461 message lines logged
Log Buffer (1000000 bytes):
Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222: //CMM/INFO:cmm_notify_trigger() 15, callID 99685, 5114016, 1884814040,
 1632257208
```

```
Jun  4 22:30:24.222: //CMM/INFO:    target_node 0
Jun  4 22:30:24.222: //CMM/INFO:Lineinfo node Search FAILED
Jun  4 22:30:24.222: //CMM/INFO:create_lineinfo_node
Jun  4 22:30:24.222: //CMM/INFO:    target_node 66AF3714
Jun  4 22:30:24.222: //CMM/INFO:   - dn 4016
Jun  4 22:30:24.222: //CMM/INFO:  CallEntry 709C3FB8
Jun  4 22:30:24.222: //CMM/INFO:  dstCallID -1
Jun  4 22:30:24.222: //CMM/INFO:  line_info 66AF3720, dn 4016
Jun  4 22:30:24.222: //CMM/INFO:    * cmm_crs_proc_tr_rpt_orig
Jun  4 22:30:24.222: //CMM/INFO:    callID = 99685, CG 5114016, GCID
=05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:increase_gcid_ref_count 99685 0
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 0
Jun  4 22:30:24.222: //CMM/INFO:      Gcidinfo node Search FAILED
Jun  4 22:30:24.222: //CMM/INFO:create_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:      count = 1
Jun  4 22:30:24.222: //CMM/INFO:insert_ssptrs_to_gcid for line_info 66AF3720 (dn 4016),
GCID 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222:     ss_ptr list :-
Jun  4 22:30:24.222:     ss_ptr list :-
Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222: //CMM/INFO:cmm_notify_trigger() 1, callID 99685, 5114016, 16, 1695547392
Jun  4 22:30:24.222: //CMM/INFO:    target_node 66AF3714
Jun  4 22:30:24.222: //CMM/INFO:   - dn 4016
Jun  4 22:30:24.222: //CMM/INFO:  CallEntry 709C3FB8
Jun  4 22:30:24.222: //CMM/INFO:  dstCallID -1
Jun  4 22:30:24.222: //CMM/INFO:  line_info 66AF3720, dn 4016
Jun  4 22:30:24.222: //CMM/INFO:    * cmm_crs_proc_tr_call_orig
Jun  4 22:30:24.222: //CMM/INFO:       orig --> callID 99685, line_info 66AF3720, call_inst
 655AF384, gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:is_sccp_endpoint DN 4016
Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222:     sccp endpoint TRUE
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:cmm_send_dialog_notify  sub_info 0
Jun  4 22:30:24.222:     ss_ptr list :-
Jun  4 22:30:24.222: //CMM/INFO:        <== DIALOG MGR ==>
Jun  4 22:30:24.222: //CMM/INFO:            :: CMM_EV_CALL_CONN_ORIGINATED
Jun  4 22:30:24.222: //CMM/INFO:                - Gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:                 - Calling 4016
Jun  4 22:30:24.222: //CMM/INFO:                 - Called
Jun  4 22:30:24.222: //CMM/INFO:                 - ConnAddr 4016
Jun  4 22:30:24.222: //CMM/INFO:                 - Type 0
Jun  4 22:30:24.222: //CMM/INFO:                 - parentGcid
00000000-00000000-00000000-00000000
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/DETAIL: type: CMM_EV_CALL_CONN_ORIGINATED, filter analyzing....
 [4016, , 4016]
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/DETAIL:gcid is not part of conference. [4016, , 4016] checking
originateFilter...
Jun  4 22:30:24.222: //CMM/DETAIL:originateFilter[callid=99685, pdn=16, pchan=1] is not
set. [4016, , 4016] is  not filtered
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
```

```
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:cmm_send_dialog_notify  sub_info 0
Jun  4 22:30:24.222:     ss_ptr list :-
Jun  4 22:30:24.222: //CMM/INFO:        <== DIALOG MGR ==>
Jun  4 22:30:24.222: //CMM/INFO:            :: CMM_EV_CALL_CONN_ACTIVE
Jun  4 22:30:24.222: //CMM/INFO:                - Gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:                 - Calling 4016
Jun  4 22:30:24.222: //CMM/INFO:                 - Called
Jun  4 22:30:24.222: //CMM/INFO:                 - ConnAddr 4016
Jun  4 22:30:24.222: //CMM/INFO:                 - LastRedirectAddr
Jun  4 22:30:24.222: //CMM/INFO:                 - Type 0
Jun  4 22:30:24.222: //CMM/INFO:                 - parentGcid
00000000-00000000-00000000-00000000
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/DETAIL: type: CMM_EV_CALL_CONN_ACTIVE, filter analyzing....
[4016, , 4016]
Jun  4 22:30:24.222: //CMM/DETAIL:called number is not specified. [4016, , 4016]
Jun  4 22:30:24.222: //CMM/DETAIL:originateFilter[callid=99685, pdn=16, pchan=1] is not
set, [4016, , 4016] is not filtered
Jun  4 22:30:25.670: //CMM/INFO:
Jun  4 22:30:25.670: //CMM/INFO:
Jun  4 22:30:25.670: //CMM/INFO:cmm_notify_trigger() 14, callID 99686, 8101, 1902058375, 0
Jun  4 22:30:25.670: //CMM/INFO:    target_node 65DB15E4
Jun  4 22:30:25.670: //CMM/INFO:   - dn 8101
.
.
.
```

Related Commands

Command	Description
callmonitor	Enable call monitoring messaging functionality on a SIP endpoint in a VoIP network.
gcid	Enable Global Call ID (Gcid) for every call on an outbound leg of a VoIP dial peer for a SIP endpoint.

debug capf-server

To collect debug information about the CAPF server, use the **debug capf-server** command in privileged EXEC mode. To disable collection of debug information, use the **no** form of this command.

debug capf-server {**all** | **error** | **events** | **messages**}
no debug capf-server

Syntax Description

all	Collect all CAPF information available.
error	Collect only information about CAPF errors.
events	Collect only information about CAPF status events.
messages	Collect only CAPF system messages.

Command Default

Collection of CAPF debug information is disabled.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CallManager Express phone authentication.

Examples

The following example shows debug messages for the CAPF server.

```
Router# debug capf-server all
001891: .Jul 21 18:17:07.014: %IPPHONE-6-UNREGISTER_NORMAL: ephone-1:SEP000E325C9A43
IP:10.10.10.194 So
cket:3 DeviceType:Phone has unregistered normally.
001892: .Jul 21 18:17:20.495: New Connection from phone, socket 1
001893: .Jul 21 18:17:20.495: Created New Handshake Process
001894: .Jul 21 18:17:20.499: SSL Handshake Error -6983
001895: .Jul 21 18:17:21.499: SSL Handshake Error -6983
001896: .Jul 21 18:17:22.555: SSL Handshake Successful
001897: .Jul 21 18:17:22.555: ephone_capf_send_auth_req:
001898: .Jul 21 18:17:22.555: ephone_capf_ssl_write: 12 bytes
001899: .Jul 21 18:17:22.711: ephone_capf_ssl_read: Read 35 bytes
001900: .Jul 21 18:17:22.711: ephone_capf_handle_phone_msg: msgtype 2
001901: .Jul 21 18:17:22.711: ephone_capf_process_auth_res_msg: SEP000E325C9A43 AuthMode 2
001902: .Jul 21 18:17:22.711: ephone_capf_send_delete_cert_req_msg: SEP000E325C9A43
001903: .Jul 21 18:17:22.711: ephone_capf_ssl_write: 8 bytes
001904: .Jul 21 18:17:23.891: ephone_capf_ssl_read: Read 12 bytes
001905: .Jul 21 18:17:23.891: ephone_capf_handle_phone_msg: msgtype 14
001906: .Jul 21 18:17:23.891: certificate delete successful for SEP000E325C9A43
001907: .Jul 21 18:17:24.695: ephone_capf_release_session: SEP000E325C9A43
001908: .Jul 21 18:17:24.695: ephone_capf_send_end_session_msg: SEP000E325C9A43
001909: .Jul 21 18:17:24.695: ephone_capf_ssl_write: 12 bytes
001910: .Jul 21 18:17:25.095: %IPPHONE-6-REG_ALARM: 22: Name=SEP000E325C9A43 Load=7.2(2.0)
```

```
 Last=Rese
t-Reset
001911: .Jul 21 18:17:25.099: %IPPHONE-6-REGISTER: ephone-1:SEP000E325C9A43 IP:10.10.10.194
 Socket:2 De
viceType:Phone has registered.
001912: .Jul 21 18:18:05.171: %IPPHONE-6-UNREGISTER_NORMAL: ephone-1:SEP000E325C9A43
IP:1.1.1.127 So
cket:2 DeviceType:Phone has unregistered normally.
001913: .Jul 21 18:18:18.288: New Connection from phone, socket 1
001914: .Jul 21 18:18:18.288: Created New Handshake Process
001915: .Jul 21 18:18:18.292: SSL Handshake Error -6983
001916: .Jul 21 18:18:19.292: SSL Handshake Error -6983
001917: .Jul 21 18:18:20.348: SSL Handshake Successful
001918: .Jul 21 18:18:20.348: ephone_capf_send_auth_req:
001919: .Jul 21 18:18:20.348: ephone_capf_ssl_write: 12 bytes^Z
001920: .Jul 21 18:18:20.492: ephone_capf_ssl_read: Read 35 bytes
001921: .Jul 21 18:18:20.492: ephone_capf_handle_phone_msg: msgtype 2
001922: .Jul 21 18:18:20.492: ephone_capf_process_auth_res_msg: SEP000E325C9A43 AuthMode 2
001923: .Jul 21 18:18:20.492: ephone_capf_send_PhKeyGenReq_msg: SEP000E325C9A43 KeySize
1024
001924: .Jul 21 18:18:20.492: ephone_capf_ssl_write: 13 bytes
001925: .Jul 21 18:18:20.540: ephone_capf_ssl_read: Read 8 bytes
001926: .Jul 21 18:18:20.540: ephone_capf_handle_phone_msg: msgtype 17
001927: .Jul 21 18:18:20.540: ephone_capf_process_req_in_progress: SEP000E325C9A43 delay
0sh
001928: .Jul 21 18:18:21.924: %SYS-5-CONFIG_I: Configured from console by user1 on console
```

debug cch323 video

To provide debugging output for video components within the H.323 subsystem, use the **debug cch323 video command in privileged EXEC mode.** To disable debugging output, use the **no** form of this command.

debug cch323 video
no debug cch323 video

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to enable a debugging trace for the video component in an H.323 network.

Examples

Originating Gateway Example

The following is sample output of the debugging log for an originating Cisco Unified CallManager Express (Cisco Unified CME) gateway after the **debug cch323 video** command was enabled:

```
Router# show log
Syslog logging: enabled (11 messages dropped, 487 messages rate-limited,
                0 flushes, 0 overruns, xml disabled, filtering disabled)
    Console logging: disabled
    Monitor logging: level debugging, 0 messages logged, xml disabled,
                     filtering disabled
    Buffer logging: level debugging, 1144 messages logged, xml disabled,
                    filtering disabled
    Logging Exception size (4096 bytes)
    Count and timestamp logging messages: disabled
    Trap logging: level informational, 1084 message lines logged
Log Buffer (6000000 bytes):
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_peer_info: Entry
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_peer_info: Have peer
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_pref_codec_list: First preferred
 codec(bytes)=16(20)
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_peer_info: Flow Mode set to
FLOW_THROUGH
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_caps_chn_info: No peer leg setup
 params
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_caps_chn_info: Setting
CCH323_SS_NTFY_VIDEO_INFO
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_h323_control_options_outgoing:
h245 sm mode = 8463
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_h323_control_options_outgoing:
h323_ctl=0x20
Jun 13 09:19:42.010: //103030/C7838B198002/H323/cch323_rotary_validate: No peer_ccb available
```

Terminating Gateway Example

The following is sample output of the debugging log for a terminating Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) gateway after the **debug cch323 video** command was enabled:

```
Router# show log
Syslog logging: enabled (11 messages dropped, 466 messages rate-limited,
                0 flushes, 0 overruns, xml disabled, filtering disabled)
    Console logging: disabled
    Monitor logging: level debugging, 0 messages logged, xml disabled,
                     filtering disabled
    Buffer logging: level debugging, 829 messages logged, xml disabled,
                    filtering disabled
    Logging Exception size (4096 bytes)
    Count and timestamp logging messages: disabled
    Trap logging: level informational, 771 message lines logged
Log Buffer (200000 bytes):
Jun 13 09:19:42.011: //103034/C7838B198002/H323/setup_ind: Receive bearer cap infoXRate 24,
 rateMult 12
Jun 13 09:19:42.011: //103034/C7838B198002/H323/cch323_set_h245_state_mc_mode_incoming:
h245 state m/c mode=0x10F, h323_ctl=0x2F
Jun 13 09:19:42.015: //-1/xxxxxxxxxxxx/H323/cch245_event_handler: callID=103034
Jun 13 09:19:42.019: //-1/xxxxxxxxxxxx/H323/cch245_event_handler: Event CC_EV_H245_SET_MODE:
 data ptr=0x465D5760
Jun 13 09:19:42.019: //-1/xxxxxxxxxxxx/H323/cch323_set_mode: callID=103034, flow Mode=1
spi_mode=0x6
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_do_call_proceeding: set_mode NOT
called yet...saved deferred CALL_PROC
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_h245_connection_sm: state=0, event=0,
 ccb=4461B518, listen state=0
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_process_set_mode: Setting inbound
leg mode flags to 0x10F, flow-mode to FLOW_THROUGH
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_process_set_mode: Sending deferred
CALL_PROC
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_do_call_proceeding: set_mode called
 so we can proceed with CALLPROC
Jun 13 09:19:42.027: //103034/C7838B198002/H323/cch323_h245_connection_sm: state=1, event=2,
 ccb=4461B518, listen state=1
Jun 13 09:19:42.027: //103034/C7838B198002/H323/cch323_send_cap_request: Setting mode to
VIDEO MODE
Jun 13 09:19:42.031: //103034/C7838B198002/H323/cch323_h245_cap_ind: Masks au=0xC data=0x2
 uinp=0x32
```

Related Commands

Command	Description
debug ephone video	Sets video debugging for the Cisco Unified IP phone.
show call active video	Displays call information for SCCP video calls in progress.
show call history video	Displays call history information for SCCP video calls.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug credentials

To set debugging on the credentials service that runs between the Cisco Unified CME CTL provider and CTL client or between the Cisco Unified SRST router and Cisco Unified CallManager, use the **debug credentials** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug credentials
no debug credentials

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.3(14)T	This command was introduced for Cisco Unified SRST.
12.4(4)XC	This command was introduced for Cisco Unified CME.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T for Cisco Unified CME.

Usage Guidelines

Cisco Unified CME

Use this command with Cisco Unified CME phone authentication to monitor a CTL provider as it provides credentials to the CTL client.

Cisco Unified SRST

Use this command to monitor Cisco Unified CallManager while it requests certificates from the Cisco Unified SRST router. It sets debugging on the credentials service that runs between the SRST router and Cisco Unified CallManager

Examples

Cisco Unified CME

The following sample output displays the CTL provider establishing a TLS session with the CTL client and providing all the relevant credentials for the services that are running on this router to the CTL client.

```
Router# debug credentials

Credentials server debugging is enabled
May 25 12:08:17.944: Credentials service: Start TLS Handshake 1 10.5.43.174 4374
May 25 12:08:17.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:18.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:19.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:20.964: Credentials service: TLS Handshake completes
```

Cisco Unified SRST

The following is sample output showing the credentials service that runs between the Cisco Unified SRST router and Cisco Unified CallManager. The credentials service provides Cisco Unified CallManager with the certificate from the SRST router.

```
Router# debug credentials
Credentials server debugging is enabled
Router#
May 25 12:08:17.944: Credentials service: Start TLS Handshake 1 10.5.43.174 4374
May 25 12:08:17.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:18.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:19.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:20.964: Credentials service: TLS Handshake completes
```

The below table describes the significant fields shown in the display.

Table 1: debug credentials Field Descriptions

Field	Description
Start TLS Handshake 1 10.5.43.174 4374	Indicates the beginning of the TLS handshake between the secure Cisco Unified SRST router and Cisco Unified CallManager. In this example, 1 indicates the socket, 10.5.43.174 is the IP address, and 4374 is the port of Cisco Unified CallManager.
TLS Handshake returns OPSSLReadWouldBlockErr	Indicates that the handshake is in process.
TLS Handshake completes	Indicates that the TLS handshake has finished and that the Cisco Unified CallManager has received the secure Cisco Unified SRST device certificate.

Related Commands

Command	Description
credentials	Enters credentials configuration mode to configure a Cisco Unified CME CTL provider certificate or a Cisco Unified SRST router certificate.
ctl-service admin	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
ip source-address (credentials)	Enables the Cisco Unified CME or SRST router to receive messages through the specified IP address and port.
show credentials	Displays the credentials settings on a Cisco Unified CME or SRST router.
show debugging	Displays information about the types of debugging that are enabled for your router.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with a Cisco Unified SRST router certificate.

debug cti

To enable debugging on the CTI interface in Cisco Unified CME, use the **debug cti** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug cti {**all** | **callcontrol** | **core** | **dmgr** | **lm** | **protoif** | **session** | **xml**}
no debug cti {**all** | **callcontrol** | **core** | **dmgr** | **lm** | **protoif** | **session** | **xml**}

Syntax Description

all	All CTI debugging traces.
callcontrol	CTI call control debugging traces.
core	Basic call debugging traces.
dmgr	CTI device manager debugging traces.
lm	CTI line monitoring debugging traces.
protoif	CTI protocol interface debugging traces.
session	CTI session degugging traces.
xml	CTI xml debugging traces.

Command Default

Debugging on the CTI interface is disabled.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.0(1)XA	This command was introduced.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command sets debugging for the CTI interface in Cisco Unified CME.

Examples

The following partial output from the **debug cti core** command shows the events from the time a call is placed to when the connection is established:

```
Router# debug cti core
Core CTI debug flags are on
.
.
.
Router#
Jun 17 23:12:09.885: //CTI/PI:cti_frontend_proc [BB5C]: received CC Event [19]:
CC_EV_CALL_INFO
Jun 17 23:12:09.885: //CTI/PI:pi_process_service_event event 19
Jun 17 23:12:09.885: //CTI/PI:    got CC_EV_CALL_INFO callID 47964
Jun 17 23:12:09.885: //CTI/PI:pi_parse_service event 0
.
.
```

```
.
Jun 17 23:12:09.889: //CTI/CC:Fsm_Idle_MakeCall calling 201, called 204
Jun 17 23:12:09.889: //CTI/DMGR:
Jun 17 23:12:09.889: MakeCall event sent to Device Manager.callID 47964, Mac:0019E83B211D,
 CallingNum:201, CalledNum:204
Jun 17 23:12:09.889: //CTI/DMGR:
Jun 17 23:12:09.889: MakeCall event sent to skinny server.Mac:0019E83B211D, CallingNum:201,
 CalledNum:204
Jun 17 23:12:09.893: //CTI/CM:-- trigger 1, callID 59291, dn 201, reason 0, result 0
Jun 17 23:12:09.893: //CTI/CM:  line_info 87674E4C, dn 201
Jun 17 23:12:09.893: //CTI/CM:    * cmm_crs_proc_tr_call_orig
Jun 17 23:12:09.893: //CTI/CM:increase_gcid_ref_count 59291 0
Jun 17 23:12:09.893: //CTI/CM:      Gcidinfo node Search FAILED
Jun 17 23:12:09.893: //CTI/CM:create_gcidinfo_node 59291
Jun 17 23:12:09.893: //CTI/CM:        orig --> callID 59291, line_info 87674E4C, call_inst
 88B0B070, gcid 1E2E3483-5ACB11DE-BA9EF925-DF2AFB55
Jun 17 23:12:09.893:  === EVENT EV_ORIGINATED
Jun 17 23:12:09.893:  201 --> . cause normal
.
.
.
Jun 17 23:12:19.217: //CTI/PI:pi_process_service_event event 20
Jun 17 23:12:19.217: //CTI/PI:    got CC_EV_CALL_INFO_ACK callID 47964
Jun 17 23:12:19.217: //CTI/SM:sm_handle_cc_service event 77
Jun 17 23:12:19.217: //CTI/SM:sm_find_scb_node_by_context context_id 47964
Jun 17 23:12:19.217: //CTI/SM:    to return 86B88298
Jun 17 23:12:19.217: //CTI/SM:    got CTI_EV_ACK, callID 47964
Jun 17 23:12:19.221: //CTI/PI:cti_frontend_proc [E750]: received CC Event [20]:
CC_EV_CALL_LOOPBACK_DONE
Jun 17 23:12:19.221: //CTI/PI:pi_process_service_event event 20
Jun 17 23:12:19.221: //CTI/PI:    got CC_EV_CALL_INFO_ACK callID 59216
Jun 17 23:12:19.221: //CTI/SM:sm_handle_cc_service event 77
Jun 17 23:12:19.221: //CTI/SM:sm_find_scb_node_by_context context_id 59216
Jun 17 23:12:19.221: //CTI/SM:    to return 87396644
Jun 17 23:12:19.221: //CTI/SM:    got CTI_EV_ACK, callID 59216
UC520#
```

Related Commands

Command	Description
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ctl-client

To collect debug information about the CTL client, use the **debug ctl-client** command in privileged EXEC configuration mode. To disable collection of debug information, use the **no** form of this command.

debug ctl-client
no debug ctl-client

Syntax Description

This command has no arguments or keywords.

Command Default

Collection of CTL client debug information is disabled.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example shows debug messages for the CTL client:

```
Router# debug ctl-client

001954: .Jul 21 18:23:02.136: ctl_client_create_ctlfile:
001955: .Jul 21 18:23:02.272: create_ctl_record: Function 0 Trustpoint cisco1
001956: .Jul 21 18:23:02.276: create_ctl_record: record added for function 0
001957: .Jul 21 18:23:02.276: create_ctl_record: Function 0 Trustpoint sast2
001958: .Jul 21 18:23:02.280: create_ctl_record: record added for function 0
001959: .Jul 21 18:23:02.280: create_ctl_record: Function 1 Trustpoint cisco1
001960: .Jul 21 18:23:02.284: create_ctl_record: record added for function 1
001961: .Jul 21 18:23:02.284: create_ctl_record: Function 3 Trustpoint cisco1
001962: .Jul 21 18:23:02.288: create_ctl_record: record added for function 3
001963: .Jul 21 18:23:02.288: create_ctl_record: Function 4 Trustpoint cisco1
001964: .Jul 21 18:23:02.292: create_ctl_record: record added for function 4
001965: .Jul 21 18:23:02.424: ctl_client_create_ctlfile: Signature length 128
001966: .Jul 21 18:23:02.640: CTL File Created Successfully
```

debug ephone alarm

To set SkinnyStation alarm messages debugging for the Cisco IP phone, use the **debug ephone alarm** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone alarm [**mac-address** *mac-address*]
no debug ephone alarm [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(8)T	This command was implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone alarm** command shows all the SkinnyStation alarm messages sent by the Cisco IP phone. Under normal circumstances, this message is sent by the Cisco IP phone just before it registers, and the message has the severity level for the alarm set to "Informational" and contains the reason for the phone reboot or re-register. This type of message is entirely benign and does not indicate an error condition.

If the **mac-address** keyword is not used, the debug ephone alarm command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following example shows a SkinnyStation alarm message that is sent before the Cisco IP phone registers:

```
Router# debug ephone alarm
phone keypad reset
CM-closed-TCP
CM-bad-state
```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone blf

To display debugging information for Busy Lamp Field (BLF) presence features, use the **debug ephone blf** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone blf [**mac-address** *mac-address*]
no debug ephone blf [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a specific IP phone.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Use this command for troubleshooting BLF speed-dial and BLF call-list features for phones in a presence service.

Examples

The following is sample output from the **debug ephone blf** command.

```
Router# debug ephone blf
EPHONE BLF debugging is enabled
*Sep  4 07:18:26.307: skinny_asnl_callback: subID 16 type 4
*Sep  4 07:18:26.307: ASNL_RESP_NOTIFY_INDICATION
*Sep  4 07:18:26.307: ephone-1[1]:ASNL notify indication message, feature index 4, subID
[16]
*Sep  4 07:18:26.307: ephone-1[1]:line status 6, subID [16]
*Sep  4 07:18:26.307: ephone-1[1]:StationFeatureStatV2Message sent, status 2
*Sep  4 07:18:26.307: skinny_asnl_callback: subID 23 type 4
*Sep  4 07:18:26.307: ASNL_RESP_NOTIFY_INDICATION
*Sep  4 07:18:26.307: ephone-2[2]:ASNL notify indication message, feature index 2, subID
[23]
*Sep  4 07:18:26.311: ephone-2[2]:line status 6, subID [23]
*Sep  4 07:18:26.311: ephone-2[2]:StationFeatureStatV2Message sent, status 2
*Sep  4 07:18:28.951: skinny_asnl_callback: subID 16 type 4
*Sep  4 07:18:28.951: ASNL_RESP_NOTIFY_INDICATION
*Sep  4 07:18:28.951: ephone-1[1]:ASNL notify indication message, feature index 4, subID
[16]
*Sep  4 07:18:28.951: ephone-1[1]:line status 1, subID [16]
*Sep  4 07:18:28.951: ephone-1[1]:StationFeatureStatV2Message sent, status 1
*Sep  4 07:18:28.951: skinny_asnl_callback: subID 23 type 4
*Sep  4 07:18:28.951: ASNL_RESP_NOTIFY_INDICATION
*Sep  4 07:18:28.951: ephone-2[2]:ASNL notify indication message, feature index 2, subID
[23]
*Sep  4 07:18:28.951: ephone-2[2]:line status 1, subID [23]
*Sep  4 07:18:28.951: ephone-2[2]:StationFeatureStatV2Message sent, status 1
```

Related Commands

Command	Description
blf-speed-dial	Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.
presence call-list	Enables BLF monitoring for call lists and directories on phones registered to a Cisco Unified CME router.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.

debug ephone ccm-compatible

To display Cisco CallManager notification updates for calls between Cisco CallManager and Cisco CallManager Express, use the **debug ephone ccm-compatible** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone ccm-compatible [**mac-address** *mac-address*]
no debug ephone ccm-compatible [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(7)T	This command was introduced.

Usage Guidelines

This command displays call flow notification information for all calls between Cisco CallManager and Cisco CallManager Express, but it is most useful for filtering out specific information for transfer and forward cases. For basic call information, use the **debug ephone state** command.

If you do not specify the **mac-address** keyword, the **debug ephone ccm-compatible** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **no** form of this command with the **mac-address** keyword.

Debugging can be enabled or disabled on any number of Cisco IP phones. Cisco IP phones that have debugging enabled are listed in the debug field of the **show ephone** command output. When debugging is enabled for a Cisco IP phone, debug output is displayed for all phone extensions (virtual voice ports) associated with that phone.

Examples

The following sample output displays call flow notifications between Cisco CallManager and Cisco CallManager Express:

```
Router# debug ephone ccm-compatible
*May  1 04:30:02.650:ephone-2[2]:DtAlertingTone/DtHoldTone - mediaActive reset during CONNECT
*May  1 04:30:02.654:ephone-2[2]:DtHoldTone - force media STOP state
*May  1 04:30:02.654://93/xxxxxxxxxxxx/CCAPI/ccCallNotify:(callID=0x5D,nData->
bitmask=0x00000007)
*May  1 04:30:02.654://93/xxxxxxxxxxxx/VTSP:(50/0/3):-1:0:5/vtsp_process_event:
 vtsp:[50/0/3 (93), S_CONNECT, E_CC_SERVICE_MSG]
*May  1 04:30:02.654://93/xxxxxxxxxxxx/VTSP:(50/0/3):-1:0:5/act_service_msg_dow
n:.
*May  1 04:30:02.658:dn_callerid_update DN 3 number= 12009 name= CCM7960 in state CONNECTED
*May  1 04:30:02.658:dn_callerid_update (incoming) DN 3 info updated to
*May  1 04:30:02.658:calling= 12009 called= 13003 origCalled=
*May  1 04:30:02.658:callingName= CCM7960, calledName= , redirectedTo =
*May  1 04:30:02.658:ephone-2[2][SEP003094C2999A]:refreshDisplayLine for line 1
 DN 3 chan 1
*May  1 04:30:03.318:ephone-2[2]:DisplayCallInfo incoming call
*May  1 04:30:03.318:ephone-2[2]:Call Info DN 3 line 1 ref 24 called 13003 calling 12009
origcalled 13003 calltype 1
*May  1 04:30:03.318:ephone-2[2]:Original Called Name UUT4PH3
```

```
*May  1 04:30:03.318:ephone-2[2]:CCM7960 calling
*May  1 04:30:03.318:ephone-2[2]:UUT4PH3
```

Related Commands

Command	Description
debug ephone state	Displays call state information.
show debugging	Displays information about the types of debugging that are enabled for your router.
show ephone	Displays information about registered Cisco IP phones.

debug ephone detail

To set detail debugging for the Cisco IP phone, use the **debug ephone detail** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone detail [**mac-address** *mac-address*]
no debug ephone detail [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone detail** command includes the error and state levels.

If the **mac-address** keyword is not used, the debug ephone detail command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of detail debugging of the Cisco IP phone with MAC address 0030.94c3.8724. The sample is an excerpt of some of the activities that takes place during call setup, connected state, active call, and the call being disconnected.

```
Router# debug ephone detail mac-address 0030.94c3.8724
Ephone detail debugging is enabled
1d04h: ephone-1[1]:OFFHOOK
.
.
1d04h: Skinny Call State change for DN 1 SIEZE
```

```
.
.
1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsOffHook
.
.
1d04h: ephone-1[1]:SetLineLamp 1 to ON
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 5
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 0
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 0
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 2
.
.
1d04h: ephone-1[1]:Store ReDial digit: 5002
.
SkinnyTryCall to 5002 instance 1
.
.
1d04h: ephone-1[1]:Store ReDial digit: 5002
1d04h: ephone-1[1]:
SkinnyTryCall to 5002 instance 1
.
.
1d04h: Skinny Call State change for DN 1 ALERTING
.
.
1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsRingOut
.
.
1d04h: ephone-1[1]:SetLineLamp 1 to ON
1d04h: SetCallInfo calling dn 1 dn 1
calling [5001] called [5002]
.
.
1d04h: ephone-1[1]: Jane calling
1d04h: ephone-1[1]: Jill
.
.
1d04h: SkinnyUpdateDnState by EFXS_RING_GENERATE
  for DN 2 to state RINGING
.
.
1d04h: SkinnyGetCallState for DN 2 CONNECTED
.
.
1d04h: ephone-1[1]:SetLineLamp 3 to ON
1d04h: ephone-1[1]:UpdateCallState DN 1 state 4 calleddn 2
.
.
1d04h: Skinny Call State change for DN 1 CONNECTED
.
.
1d04h: ephone-1[1]:OpenReceive DN 1 codec 4:G711Ulaw64k  duration 10 ms bytes 80
.
.
1d04h: ephone-1[1]:OpenReceiveChannelAck 1.2.172.21 port=20180
1d04h: ephone-1[1]:Outgoing calling DN 1 Far-ephone-2 called DN 2
```

```
1d04h: SkinnyGetCallState for DN 1 CONNECTED
.
.
1d04h: ephone-1[1]:SetCallState line 3 DN 2 TsOnHook
.
.
1d04h: ephone-1[1]:SetLineLamp 3 to OFF
.
.
1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsOnHook
.
.
1d04h: ephone-1[1]:Clean Up Speakerphone state
1d04h: ephone-1[1]:SpeakerPhoneOnHook
1d04h: ephone-1[1]:Clean up activeline 1
1d04h: ephone-1[1]:StopTone sent to ephone
1d04h: ephone-1[1]:Clean Up phone offhook state
1d04h: SkinnyGetCallState for DN 1 IDLE
1d04h: called DN -1, calling DN -1 phone -1
1d04h: ephone-1[1]:SetLineLamp 1 to OFF
1d04h: UnBinding ephone-1 from DN 1
1d04h: UnBinding called DN 2 from DN 1
1d04h: ephone-1[1]:ONHOOK
1d04h: ephone-1[1]:SpeakerPhoneOnHook
1d04h: ephone-1[1]:ONHOOK NO activeline
.
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone error

To set error debugging for the Cisco IP phone, use the **debug ephone error** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone error [**mac-address** *mac-address*]
no debug ephone error [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone error** command cancels debugging at the detail and state level.

If the **mac-address** keyword is not used, the debug ephone error command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of error debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone error mac-address 0030.94c3.8724
EPHONE error debugging is enabled
socket [2] send ERROR 11
Skinny Socket [2] retry failure
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone extension-assigner

To display status messages produced by the extension assigner application, use the **debug ephone extension-assigner** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone extension-assigner
no debug ephone extension-assigner

Syntax Description

This command has no arguments or keywords.

Command Default

Debug ephone extension-assigner is disabled.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command displays status messages produced by the extension assigner application, including messages related to the functions performed by the following Tcl commands:

- phone query—Verifies whether the ephone tag has been assigned a MAC address.
- phone assign—Binds the MAC address from the caller's phone to a preexisting ephone template.
- phone unassign—Removes the MAC address from the ephone tag.

Before using this command, you must load the Tcl script for the extension assigner application.

Examples

The following is sample output of extension assigner debugging as the extension assigner application queries phones for their status and issues commands to assign or unassign extension numbers.

```
*Jun 9 19:08:10.627: ephone_query: inCallID=47, tag=4, ephone_tag=4
*Jun 9 19:08:10.627: extAssigner_IsEphoneMacPreset: ephone_tag = 4, ipKeyswitch.max_ephones
 = 96
*Jun 9 19:08:10.627: extAssigner_IsEphoneMacPreset: ephone_ptr->mac_addr_str = 000B46BDE075,
 MAC_EXT_RESERVED_VALUE = 02EAEAEA0000
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: callID = 47
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->physical_interface_type
(26); CV_VOICE_EFXS (26)
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->type (6); CC_IF_TELEPHONY
 (6)
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: htsp->sig_type (26); CV_VOICE_EFXS
 (26)
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: dn = 4, chan = 1
*Jun 9 19:08:10.627: ephone_query: EXTASSIGNER_RC_SLOT_ASSIGNED_TO_CALLING_PHONE
*Jun 9 19:08:22.763: ephone_unassign: inCallID=47, tag=4, ephone_tag=4
*Jun 9 19:08:22.763: extAssigner_IsEphoneMacPreset: ephone_tag = 4, ipKeyswitch.max_ephones
 = 96
```

```
*Jun 9 19:08:22.763: extAssigner_IsEphoneMacPreset: ephone_ptr->mac_addr_str = 000B46BDE075,
 MAC_EXT_RESERVED_VALUE = 02EAEAEA000
*Jun 9 19:08:22.763: is_ephone_auto_assigned: button-1 dn_tag=4
*Jun 9 19:08:22.763: is_ephone_auto_assigned: NO
*Jun 9 19:08:22.763: SkinnyGetActivePhoneIndexFromCallid: callID = 47
*Jun 9 19:08:22.763: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->physical_interface_type
(26); CV_VOICE_EFXS (26)
*Jun 9 19:08:22.767: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->type (6); CC_IF_TELEPHONY
 (6)
*Jun 9 19:08:22.767: SkinnyGetActivePhoneIndexFromCallid: htsp->sig_type (26); CV_VOICE_EFXS
 (26)
*Jun 9 19:08:22.767: SkinnyGetActivePhoneIndexFromCallid: dn = 4, chan = 1
*Jun 9 19:08:29.795: ephone-4[8]:fStationOnHookMessage: Extension Assigner request restart,
 cmd=2, new mac=02EAEAEA0004, ephone_tag=4
*Jun 9 19:08:30.063: %IPPHONE-6-UNREGISTER_NORMAL: ephone-4:SEP000B46BDE075 IP:5.5.0.1
Socket:8 DeviceType:Phone has unregistered normally.
*Jun 9 19:08:30.063: ephone-4[8][SEP000B46BDE075]:extAssigner_assign: new mac=02EAEAEA0004,
 ephone-tag=4
*Jun 9 19:08:30.063: extAssigner_simple_assign: mac=02EAEAEA0004, tag=4
*Jun 9 19:08:30.063: ephone_updateCNF: update cnf_file ephone_tag=4
*Jun 9 19:08:30.063: extAssigner_assign: restart again (mac=02EAEAEA0004) ephone_tag=4
*Jun 9 19:08:30.131: %IPPHONE-6-REG_ALARM: 23: Name=SEP000B46BDE075 Load=8.0(2.0)
Last=Reset-Restart
*Jun 9 19:08:30.135: %IPPHONE-6-REGISTER_NEW: ephone-7:SEP000B46BDE075 IP:5.5.0.1 Socket:10
 DeviceType:Phone has registered.
*Jun 9 19:08:30.503: %IPPHONE-6-UNREGISTER_NORMAL: ephone-7:SEP000B46BDE075 IP:5.5.0.1
Socket:10 DeviceType:Phone has unregistered normally.
*Jun 9 19:08:43.127: %IPPHONE-6-REG_ALARM: 22: Name=SEP000B46BDE075 Load=8.0(2.0)
Last=Reset-Reset
*Jun 9 19:08:43.131: %IPPHONE-6-REGISTER: ephone-7:SEP000B46BDE075 IP:5.5.0.1 Socket:13
DeviceType:Phone has registered.
```

Command	Description
debug ephone state	Sets state debugging for Cisco IP phones.
debug voip application script	Displays status messages produced by voice over IP application scripts.

debug ephone hfs

To collect and display debugging information on the download of IP phone configuration and firmware files using the HTTP File-Fetch Server (HFS) service in a Cisco Unified CME system, use the **debug ephone hfs** command in privileged EXEC mode. To disable collection of debug information, use the **no** form of this command.

[**no**] **debug ephone hfs**

Syntax Description

This command has no arguments or keywords.

Command Default

There are no debug logs on the console or buffer.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

Use the **debug ephone hfs** command to troubleshoot an attempt to download Cisco Unified SIP IP phone configuration and firmware files using the HFS service.

Examples

The following sample display shows a successful file fetch:

```
Router# debug ephone hfs
Jan  5 01:29:00.829: ephone_hfs_util_urlhook:URL Context --->
    svr_port=6970
    rem_port=63881
    is_ssl=0
    req_method=1
    url=/softkeyDefault.xml
Jan  5 01:29:00.833: ephone_hfs_util_urlhook:Found the binding, fn[softkeyDefault.xml],
path[system:/ephone/sipphone/softkeyDefault.xml]
Jan  5 01:29:00.833: ephone_hfs_util_get_action:Get HTTP-url[/softkeyDefault.xml],
fetch_path[system:/ephone/sipphone/softkeyDefault.xml], fetch_from_home[0]
Jan  5 01:29:00.853: HFS SUCCESS !!! fn=system:/ephone/sipphone/softkeyDefault.xml size=4376
 upload-time(s.ms)=0.016
```

The following sample display shows an unsuccessful file fetch, where the file is not found:

```
Router# debug ephone hfs
Jan  5 01:43:16.561: ephone_hfs_util_urlhook:URL Context --->
    svr_port=6970
    rem_port=63890
    is_ssl=0
    req_method=1
    url=/softkeyDefault2.xml
Jan  5 01:43:16.561: ephone_hfs_util_urlhook:File not found
```

The table describes the significant fields shown in the display.

Table 2: debug ephone hfs Field Descriptions

Field	Description
svr_port	Cisco Unified CME port where the request is sent by the remote Cisco Unified SIP IP phone.
rem_port	Remote port of the Cisco Unified SIP IP phone. The request originates from this port.
is_ssl	Indicates if a secure HTTP connection is established using the Secure Sockets Layer (SSL) method.
req_method	Indicates the type of HTTP request message. A value of 1 is equivalent to HTTP-GET while a value of 2 is equivalent to HTTP-POST.
url	Location of the file to be downloaded.

Related Commands

Command	Description
hfs enable	Enables the HFS download service on an IP Phone in a Cisco Unified CME system.

debug ephone keepalive

To set keepalive debugging for the Cisco IP phone, use the **debug ephone keepalive** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone keepalive [**mac-address** *mac-address*]
no debug ephone keepalive [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone keepalive** command sets keepalive debugging.

If the **mac-address** keyword is not used, the debug ephone keepalive command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of the keepalive status for the Cisco IP phone with MAC address 0030.94C3.E1A8:

```
Router# debug ephone keepalive mac-address 0030.94c3.E1A8
EPHONE keepalive debugging is enabled for phone 0030.94C3.E1A8
1d05h: ephone-1 Set interface FastEthernet0/0  ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: ephone-1 Set interface FastEthernet0/0  ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: Skinny Checking for stale sockets
1d05h: ephone-1 Set interface FastEthernet0/0  ETHERNET
```

```
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: ephone-1 Set interface FastEthernet0/0  ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: Skinny active socket list (3/96):  1 2 4
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone loopback

To set debugging for loopback calls, use the **debug ephone loopback** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone loopback [**mac-address** *mac-address*]
no debug ephone loopback [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced for Cisco IOS Telephony Services (now known as Cisco CallManager Express) Version 2.0 on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The **debug ephone loopback** command sets debugging for incoming and outgoing calls on all loopback-dn pairs or on the single loopback-dn pair that is associated with the IP phone that has the MAC address specified in this command.

If you enable the **debug ephone loopback** command and the **debug ephone pak** command at the same time, the output displays packet debug output for the voice packets that are passing through the loopback-dn pair.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with that Cisco IP phone.

Examples

The following example contains two excerpts of output for a call that is routed through a loopback. The first excerpt is output from the **show running-config** command and displays the loopback configuration used for this example. The second excerpt is output from the **debug ephone loopback** command.

```
Router# show running-config
.
.
.
ephone-dn 14
 number 1514
!
!
ephone-dn  42
```

```
 number 17181..
 loopback-dn 43 forward 4
 no huntstop
!
!
ephone-dn  43
 number 19115..
 loopback-dn 42 forward 4
!
.
.
.
```

A loopback call is started. An incoming call to 1911514 (ephone-dn 43) uses the loopback pair of ephone-dns to become an outgoing call to extension 1514. The number in the outgoing call has only four digits because the **loopback-dn** command specifies forwarding of four digits. The outgoing call uses ephone-dn 42, which is also specified in the **loopback-dn** command under ephone-dn 43. When the extension at 1514 rings, the following debug output is displayed:

```
Router# debug ephone loopback
Mar  7 00:57:25.376:Pass processed call info to special DN 43 chan 1
Mar  7 00:57:25.376:SkinnySetCallInfoLoopback DN 43 state IDLE to DN 42 state IDLE
Mar  7 00:57:25.376:Called Number = 1911514 Called Name =
Mar  7 00:57:25.376:Calling Number = 8101 Calling Name =
 orig Called Number =
Copy Caller-ID info from Loopback DN 43 to DN 42
Mar  7 00:57:25.376:DN 43  Forward 1514
Mar  7 00:57:25.376:PredictTarget match 1514 DN 14 is idle
Mar  7 00:57:25.380:SkinnyUpdateLoopbackState DN 43 state RINGING calledDn -1
Mar  7 00:57:25.380:Loopback DN 42 state IDLE
Mar  7 00:57:25.380:Loopback DN 43 calledDN -1 callingDn -1 G711Ulaw64k
Mar  7 00:57:25.380:SkinnyUpdateLoopbackState DN 43 to DN 42 signal OFFHOOK
Mar  7 00:57:25.380:SetDnCodec Loopback DN 43 codec 4:G711Ulaw64k  vad 0 size 160
Mar  7 00:57:25.380:SkinnyDnToneLoopback DN 42 state SIEZE to DN 43 state RINGING
Mar  7 00:57:25.380:TONE ON DtInsideDialTone
Mar  7 00:57:25.380:SkinnyDnToneLoopback called number = 1911514
Mar  7 00:57:25.380:DN 43  Forward 1514
Mar  7 00:57:25.380:DN 42 from 43  Dial 1514
Mar  7 00:57:25.384:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
Mar  7 00:57:25.384:TONE OFF
Mar  7 00:57:25.384:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
Mar  7 00:57:25.384:TONE OFF
Mar  7 00:57:25.384:SkinnyUpdateLoopbackState DN 42 state ALERTING calledDn -1
Mar  7 00:57:25.384:Loopback DN 43 state RINGING
Mar  7 00:57:25.384:Loopback Alerting DN 42 calledDN -1 callingDn -1 G711Ulaw64k
Mar  7 00:57:25.388:ephone-5[7]:DisplayCallInfo incoming call
Mar  7 00:57:25.388:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
Mar  7 00:57:25.388:TONE ON DtAlertingTone
Mar  7 00:57:25.388:SkinnyDnToneLoopback DN 42 to DN 43 deferred alerting by DtAlertingTone
Mar  7 00:57:25.388:EFXS_STATE_ONHOOK_RINGING already done for DN 43 chan 1
Mar  7 00:57:25.388:Set prog_ind 0 for DN 42 chan 1
.
.
.
```

When extension 1514 answers the call, the following debug output is displayed:

```
.
.
.
Mar  7 00:57:32.158:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
Mar  7 00:57:32.158:TONE OFF
```

```
Mar  7 00:57:32.158:dn_support_g729 true DN 42 chan 1 (loopback)
Mar  7 00:57:32.158:SetDnCodec Loopback DN 43 codec 4:G711Ulaw64k  vad 0 size 160
Mar  7 00:57:32.158:SkinnyUpdateLoopbackState DN 42 state CALL_START calledDn 14
Mar  7 00:57:32.158:Loopback DN 43 state RINGING
Mar  7 00:57:32.158:SkinnyUpdateLoopbackState DN 42 to DN 43 deferred alerting by CALL_START
 already sent
Mar  7 00:57:32.158:SetDnCodec reassert defer_start for DN 14 chan 1
Mar  7 00:57:32.158:Delay media until loopback DN 43 is ready
Mar  7 00:57:32.158:SkinnyUpdateLoopbackCodec check for DN 14 chan 1 from DN 42 loopback
DN 43
Mar  7 00:57:32.158:SkinnyUpdateLoopbackCodec DN chain is 14 1, other=42, lb=43, far=-1 1,
 final=43 1
Mar  7 00:57:32.158:SkinnyUpdateLoopbackCodec DN 14 chan 1 DN 43 chan 1 codec 4 match
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 42 state CONNECTED calledDn 14
Mar  7 00:57:32.162:Loopback DN 43 state RINGING
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 42 to DN 43 signal ANSWER
Mar  7 00:57:32.162:Loopback DN 42 calledDN 14 callingDn -1 G711Ulaw64k
Mar  7 00:57:32.162:Loopback DN 43 calledDN -1 callingDn -1 incoming G711Ulaw64k
Mar  7 00:57:32.162:ephone-5[7][SEP000DBDBEF37D]:refreshDisplayLine for line 1 DN 14 chan
1
Mar  7 00:57:32.162:dn_support_g729 true DN 43 chan 1 (loopback)
Mar  7 00:57:32.162:SetDnCodec Loopback DN 42 codec 4:G711Ulaw64k  vad 0 size 160
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 43 state CALL_START calledDn -1
Mar  7 00:57:32.162:Loopback DN 42 state CONNECTED
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 43 has defer_dn 14 chan 1 set
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 43 has defer_dn 14 chan 1:
 -invoke SkinnyOpenReceive
Mar  7 00:57:32.162:SkinnyUpdateLoopbackCodec check for DN 14 chan 1 from DN 42 loopback
DN 43
Mar  7 00:57:32.162:SkinnyUpdateLoopbackCodec DN chain is 14 1, other=42, lb=43, far=-1 1,
 final=43 1
Mar  7 00:57:32.162:SkinnyUpdateLoopbackCodec DN 14 chan 1 DN 43 chan 1 codec 4 match
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 43 state CALL_START calledDn -1
Mar  7 00:57:32.162:Loopback DN 42 state CONNECTED
Mar  7 00:57:32.454:SkinnyGetDnAddrInfo DN 43 LOOPBACK
update media address to 10.0.0.6 25390 from DN 14
Mar  7 00:57:33.166:ephone-5[7]:DisplayCallInfo incoming call
.
.
.
```

When the called extension, 1514, goes back on-hook, the following debug output is displayed:

```
.
.
.
Mar  7 00:57:39.224:Loopback DN 42 disc reason 16 normal state CONNECTED
Mar  7 00:57:39.224:SkinnyUpdateLoopbackState DN 42 state CALL_END calledDn -1
Mar  7 00:57:39.224:Loopback DN 43 state CONNECTED
Mar  7 00:57:39.224:SkinnyUpdateLoopbackState DN 42 to DN 43 signal ONHOOK
Mar  7 00:57:39.236:SkinnyDnToneLoopback DN 42 state IDLE to DN 43 state IDLE
Mar  7 00:57:39.236:TONE OFF
Mar  7 00:57:39.236:SkinnyDnToneLoopback DN 43 state IDLE to DN 42 state IDLE
Mar  7 00:57:39.236:TONE OFF
```

The below table describes the significant fields shown in the display.

Table 3: debug ephone loopback Field Descriptions

Field	Description
Called Number	Original called number as presented to the incoming side of the loopback-dn.

Field	Description
Forward	Outgoing number that is expected to be dialed by the outgoing side of the loopback-dn pair.
PredictTarget Match	Extension (ephone-dn) that is anticipated by the loopback-dn to be the far-end termination for the call.
signal OFFHOOK	Indicates that the outgoing side of the loopback-dn pair is going off-hook prior to placing the outbound call leg.
Dial	Outbound side of the loopback-dn that is actually dialing the outbound call leg.
deferred alerting	Indicates that the alerting, or ringing, tone is returning to the original inbound call leg in response to the far-end ephone-dn state.
DN chain	Chain of ephone-dns that has been detected, starting from the far-end that terminates the call. Each entry in the chain indicates an ephone-dn tag and channel number. Entries appear in the following order, from left to right: • Ephone-dn tag and channel of the far-end call terminator (in this example, ephone-dn 14 is extension 1514). • other—Ephone-dn tag of the outgoing side of the loopback. • lb—Ephone-dn tag of the incoming side of the loopback. • far—Ephone-dn tag and channel of the far-end call originator, or -1 for a nonlocal number. • final—Ephone-dn tag for the originator of the call on the incoming side of the loopback. If the originator is not a local ephone-dn, this is set to -1. This number represents the final ephone-dn tag in the chain, looking toward the originator.
codec match	Indicates that there is no codec conflict between the two calls on either side of the loopback-dn.
GetDnAddrInfo	IP address of the IP phone at the final destination extension (ephone-dn), after resolving the chain of ephone-dns involved.
disc_reason	Disconnect cause code, in decimal. These are normal CC_CAUSE code values that are also used in call control API debugging. Common cause codes include the following: • 16—Normal disconnect. • 17—User busy. • 19—No answer. • 28—Invalid number.

Related Commands

Command	Description
debug ephone pak	Provides voice packet level debugging.
loopback-dn	Configures loopback-dn virtual loopback voice ports used to establish demarcation points for VoIP voice calls and supplementary services.

Command	Description
show ephone	Displays information about registered Cisco IP phones.
show ephone-dn loopback	Displays information for ephone-dns that have been set up for loopback calls.

debug ephone lpcor

To display debugging information for calls using the logical partitioning class of restriction (LPCOR) feature, use the **debug ephone lpcor** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone lpcor [**mac-address** *mac-address*]
no debug ephone lpcor [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a specific IP phone.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.0(1)XA	This command was introduced.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use this command for troubleshooting LPCOR calls to phones in a Cisco Unified CME system.

If the **mac-address** keyword is not used, this command debugs all phones that are registered to the Cisco Unified CME router. You can disable debugging for specific phones by using the **mac-address** keyword with the **no** form of this command.

Examples

The following is sample output from the **debug ephone lpcor** command for a call between ephone-1 and ephone-2 that was blocked by LPCOR policy validation:

```
Router# debug ephone lpcor
*Jun 24 11:23:45.599: ephone-1[0/3][SEP003094C25F38]:ephone_get_lpcor_index: dir 0
*Jun 24 11:23:46.603: ephone-2[1/2][SEP0021A02DB62A]:ephone_get_lpcor_index: dir 1
```

Related Commands

Command	Description
debug voip application lpcor	Enables debugging of the LPCOR application system.
debug voip lpcor	Displays debugging information for the LPCOR feature.
lpcor incoming	Associates an incoming call with a LPCOR resource-group policy.
lpcor outgoing	Associates an outgoing call with a LPCOR resource-group policy.
show ephone	Displays information about phones registered to Cisco Unified CME.
show voice lpcor policy	Displays the LPCOR policy for the specified resource group.

debug ephone message

To enable message tracing between ephones, use the **debug ephone message** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone message [**detail**]
no debug ephone message

Syntax Description

detail	(Optional) Displays signaling connection control protocol (SCCP) messages sent and received between ephones in the Cisco Unified CallManager Express (Cisco Unified CME) system.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **debug ephone message** command enables message tracing between ephones.

The debug ephone command debugs all ephones associated with a Cisco Unified CME router.

You can enable or disable debugging on any number of ephones. To see the ephones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a ephone, the debug output is displayed for the directory numbers associated with the ephone.

Examples

The following is sample output for the **debug ephone message** command for ephones:

```
Router# debug ephone message
EPHONE skinny message debugging is enabled
*Jul 17 12:12:54.883: Received message from phone 7, SkinnyMessageID = StationKe
epAliveMessageID
*Jul 17 12:12:54.883: Sending  message to   phone 7, SkinnyMessageID = StationKe
epAliveAckMessageID
```

The following command disables ephone message debugging:

```
Router# no debug ephone message
EPHONE skinny message debugging is disabled
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the ephone.
debug ephone detail	Sets detail debugging for the ephone.
debug ephone error	Sets error debugging for the ephone.
debug ephone mwi	Sets MWI debugging for the ephone.

Command	Description
debug ephone pak	Provides voice packet level debugging and displays the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the ephone.
debug ephone state	Sets state debugging for the ephone.
debug ephone statistics	Sets statistics debugging for the ephone.
debug ephone video	Sets video debugging for the ephone.
show debugging	Displays information about the types of debugging that are enabled for your router.
show ephone	Displays information about ephones.

debug ephone mlpp

To display debugging information for Multilevel Precedence and Preemption (MLPP) calls to phones in a Cisco Unified CME system, use the **debug ephone mlpp** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone mlpp [**mac-address** *mac-address*]
no debug ephone mlpp [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a specific IP phone.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(22)YB	This command was introduced.
12.4(24)T	This command was integrated into Cisco IOS Release 12.4(24)T .

Usage Guidelines

Use this command to troubleshoot calls that use the MLPP service.

Examples

The following is sample output from the **debug ephone mlpp** command. This example shows output for the following call scenario:

- Ephone 1 is connected to ephone 3 (nonMLPP call).
- Ephone 4 makes an MLPP call to ephone 3. Preemption tone is played to both ephone 1 and 3.
- Ephone 3 is disconnected after the preemption tone timeout and precedence ringing.
- Ephone 3 answers the MLPP call and is connected to ephone 4.

```
Router# debug ephone mlpp

Sep  5 14:23:00.499: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep  5 14:23:00.499: ephone-4[3/3][SEP001AE2BC3EE7]:max precedence=0
Sep  5 14:23:02.299: ephone-4[3/3][SEP001AE2BC3EE7]:mlpp_ephone_display_update callID=294
Sep  5 14:23:02.299: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep  5 14:23:02.299: ephone-4[3/3][SEP001AE2BC3EE7]:mlpp precedence=4, domain=0
Sep  5 14:23:02.303: ephone-3[2/1][SEP001B54BA0D64]:preemption=1
Sep  5 14:23:02.303: ephone-3[2/1][SEP001B54BA0D64]:preemption=1
Sep  5 14:23:02.303: mlpp_ephone_find_call: preempt_htsp=1774234732,
prempt_htsp->mlpp_preemptor_cid=294
Sep  5 14:23:02.303: //294/A6B5C03A8141/VOIP-MLPP/voice_mlpp_get_preemptInfo:
   mlpp_ephone_find_call is successful
Sep  5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep  5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:mlpp precedence=4, domain=0
Sep  5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep  5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:mlpp precedence=4, domain=0
Sep  5 14:23:02.303: ephone-6[5/6][SEP0018187F49FD]:indication=1
Sep  5 14:23:02.303: ephone-6[5/6][SEP0018187F49FD]:mlpp precedence=4, domain=0
Sep  5 14:23:02.303: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep  5 14:23:02.307: ephone-1[0/2][SEP0014A9818797]:indication=1
Sep  5 14:23:02.307: ephone-3[2/1][SEP001B54BA0D64]:indication=1
Sep  5 14:23:02.307: ephone-1[0/2][SEP0014A9818797]:indication=1DtPreemptionTone
```

```
Sep  5 14:23:02.307: ephone-3[2/1][SEP001B54BA0D64]:indication=1DtPreemptionTone
Sep  5 14:23:07.307: ephone-3[2/1][SEP001B54BA0D64]:indication=1
Sep  5 14:23:07.307: ephone-1[0/2][SEP0014A9818797]:indication=1
Sep  5 14:23:07.319: ephone-3[2/1][SEP001B54BA0D64]:indication=1
Sep  5 14:23:07.319: ephone-3[2/1][SEP001B54BA0D64]:indication=1
Sep  5 14:23:07.319: ephone-3[2/1][SEP001B54BA0D64]:mlpp precedence=4, domain=0
Sep  5 14:23:07.319: ephone-3[2/1][SEP001B54BA0D64]:indication=1
Sep  5 14:23:07.319: ephone-3[2/1][SEP001B54BA0D64]: MLPP Precedence Ring 6 instead
Sep  5 14:23:10.623: ephone-3[2/1][SEP001B54BA0D64]:indication=1
Sep  5 14:23:10.623: ephone-3[2/1][SEP001B54BA0D64]:indication=1
Sep  5 14:23:10.623: ephone-3[2/1][SEP001B54BA0D64]:mlpp precedence=4, domain=0
Sep  5 14:23:10.623: ephone-3[2/1][SEP001B54BA0D64]:indication=1
Sep  5 14:23:10.623: ephone-3[2/1][SEP001B54BA0D64]:mlpp precedence=4, domain=0
Sep  5 14:23:10.623: ephone-4[3/3][SEP001AE2BC3EE7]:indication=1
Sep  5 14:23:10.623: ephone-4[3/3][SEP001AE2BC3EE7]:mlpp precedence=4, domain=0
Sep  5 14:23:10.623: ephone-6[5/6][SEP0018187F49FD]:indication=1
Sep  5 14:23:10.623: ephone-6[5/6][SEP0018187F49FD]:mlpp precedence=4, domain=0
```

Related Commands

Command	Description
debug voice mlpp	Displays debugging information for MLPP service.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
mlpp max-precedence	Sets the maximum precedence (priority) level that a phone user can specify when making an MLPP call.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.

debug ephone moh

To set debugging for music on hold (MOH), use the **debug ephone moh** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone moh [**mac-address** *mac-address*]
no debug ephone moh [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced for Cisco IOS Telephony Services (now known as Cisco CallManager Express) Version 2.0 and Cisco Survivable Remote Site Telephony (SRST) Version 2.0 on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Always use the **no moh** command before modifying or replacing the MOH file in Flash memory.

When a configuration using the **multicast moh** command is used and the **debug ephone moh** command is enabled, if you delete or modify the MOH file in the router's Flash memory, the debug output can be excessive and can flood the console. The multicast MOH configuration should be removed before using the **no moh** command when the **debug ephone moh** command is enabled.

Examples

The following sample output shows MOH activity prior to the first MOH session. Note that if you enable multicast MOH, that counts as the first session.

```
Router# debug ephone moh
Mar  7 00:52:33.817:MOH AU file
Mar  7 00:52:33.817:skinny_open_moh_play set type to 3
Mar  7 00:52:33.825: 2E73 6E64 0000 0018 0007 3CCA 0000 0001
Mar  7 00:52:33.825: 0000 1F40 0000 0001 FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
```

```
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825:
Mar  7 00:52:33.825:AU file processing Found .snd
Mar  7 00:52:33.825:AU file data start at 24 end at 474338
Mar  7 00:52:33.825:AU file codec Media_Payload_G711Ulaw64k
Mar  7 00:52:33.825:MOH read file header type AU start 24 end 474338
Mar  7 00:52:33.825:MOH pre-read block 0 at write-offset 0 from 24
Mar  7 00:52:33.833:MOH pre-read block 1 at write-offset 8000 from 8024
Mar  7 00:52:33.845:Starting read server with play-offset 0 write-offset 16000
```

The below table describes the significant fields shown in the display.

Table 4: debug ephone moh Field Descriptions

Field	Description
type	0—invalid 1—raw file 2—wave format file (.wav) 3—AU format (.au) 4—live feed
AU file processing Found .snd	A .snd header was located in the AU file.
AU file data start at, end at	Data start and end file offset within the MOH file, as indicated by the file header.
read file header type	File format found (AU, WAVE, or RAW).
pre-read block, write-offset	Location in the internal MOH buffer to which data is being written, and location from which that data was read in the file.
play-offset, write-offset	Indicates the relative positioning of MOH file read-ahead buffering. Data is normally written from a Flash file into the internal circular buffer, ahead of the location from which data is being played or output.

Related Commands

Command	Description
moh (telephony-service)	Generates an audio stream from a file for MOH in a Cisco CME system.
multicast moh	Uses the MOH audio stream as a multicast source in a Cisco CME system.

debug ephone mwi

To set message waiting indication (MWI) debugging for the Cisco IOS Telephony Service router, use the **debug ephone mwi** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone mwi
no debug ephone mwi

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(8)T	This command was implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone mwi** command sets message waiting indication debugging for the Cisco IOS Telephony Service router. Because the MWI protocol activity is not specific to any individual Cisco IP phone, setting the MAC address keyword qualifier for this command is not useful.

Note Unlike the other related **debug ephone** commands, the **mac-address** keyword does not help debug a particular Cisco IP phone.

Examples

The following is sample output of the message waiting indication status for the Cisco IOS Telephony Service router:

```
Router# debug ephone mwi
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.

Command	Description
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone paging

To collect debugging information on paging for both Cisco Unified SIP IP and Cisco Unified SCCP IP phones, use the **debug ephone paging** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

[**no**] **debug ephone paging**

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Examples

The following example shows debug messages from the **debug ephone paging** command:

```
*Dec  7 21:53:42.519: Paging-dn 250 sccp count=1 sip count=2
*Dec  7 21:53:42.527: SkinnyBuildPagingList for DN 250
*Dec  7 21:53:42.527: SkinnySetPagingList added DN 251 to list for DN 250
*Dec  7 21:53:42.527: SkinnySetPagingList added DN 252 to list for DN 250
*Dec  7 21:53:42.527: Paging Group List: 251 252 0 0 0 0 0 0 0 0
*Dec  7 21:53:42.527: SkinnySetupPagingDnMulticast 239.1.1.0 20480 for DN 250
*Dec  7 21:53:42.527: Found paging DN 250 on ephone-2
*Dec  7 21:53:42.527: Added interface GigabitEthernet0/0 to multicast list for DN 250
*Dec  7 21:53:42.527: SkinnyStartPagingPhone 1 for DN 250 with multicast
*Dec  7 21:53:42.527: Found paging DN 250 on pool 1[40001] is_paging=FALSE
*Dec  7 21:53:42.527: SipPagingPhoneReq for pool 1[40001] with multicast start
*Dec  7 21:53:42.527: Found paging DN 250 on pool 2[40003] is_paging=FALSE
*Dec  7 21:53:42.527: SipPagingPhoneReq for pool 2[40003] with multicast start
*Dec  7 21:53:42.531: SkinnyBuildPagingList DN 250 for 1 targets
*Dec  7 21:53:42.531: SkinnyStartPagingMedia for 1 targets for DN 250
*Dec  7 21:53:57.471: SkinnyStopPagingPhone 1 for DN 250 with multicast
*Dec  7 21:53:57.471: SipPagingPhoneReq for pool 1[40001] with multicast stop
*Dec  7 21:53:57.471: SipPagingPhoneReq for pool 2[40003] with multicast stop
```

The following example shows another set of debug messages from the **debug ephone paging** command:

```
*Oct 27 22:39:32.543: Paging-dn 251   sccp count 1   sip count 1
*Oct 27 22:39:32.551: SkinnyBuildPagingList for DN 251
*Oct 27 22:39:32.551: SkinnySetupPagingDnMulticast 239.1.1.1 20480 for DN 251
*Oct 27 22:39:32.551: Found paging DN 251 on ephone-2
*Oct 27 22:39:32.551: Added interface GigabitEthernet0/0 to multicast list for DN 251
*Oct 27 22:39:32.551: SkinnyStartPagingPhone for DN 251 with multicast
*Oct 27 22:39:32.551: Found paging DN 251 on pool 3[40007]
*Oct 27 22:39:32.551: SipPagingPhoneReq for pool 3[40007] with multicast start
*Oct 27 22:39:32.551: SkinnyBuildPagingList DN 251 for 1 targets
*Oct 27 22:39:32.551: SkinnyStartPagingMedia for 1 targets for DN 251
*Oct 27 22:39:38.331: SkinnyStopPagingPhone for DN 251 with multicast
*Oct 27 22:39:38.331: SipPagingPhoneReq for pool 3[40007] with multicast stop
```

Related Commands

Command	Description
paging-dn	Creates a paging extension to receive audio pages on a Cisco Unified IP phone in a Cisco Unified CME system.
paging-dn (voice register)	Registers a Cisco Unified SIP IP phone to an ephone-dn paging directory number.

debug ephone pak

To provide voice packet level debugging and to print the contents of one voice packet in every 1024 voice packets, use the **debug ephone pak** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone pak [**mac-address** *mac-address*]
no debug ephone pak [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone pak** command provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.

If the **mac-address** keyword is not used, the debug ephone pak command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of packet debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone pak mac-address 0030.94c3.8724
EPHONE packet debugging is enabled for phone 0030.94c3.8724
01:29:14: ***ph_xmit_ephone DN 3 tx_pkts 5770 dest=10.2.1.1 orig len=32
 pakcopy=0 discards 27 ip_enctype 0 0 last discard: unsupported payload type
```

```
01:29:14: to_skinny_duration 130210 offset -30 last -40 seq 0 adj 0
01:29:14: IP:   45B8 003C 0866 0000 3F11 3F90 2800 0001 0A02 0101
01:29:14: TTL 63 TOS B8 prec 5
01:29:14: UDP:  07D0 6266 0028 0000
01:29:14: sport 2000 dport 25190 length 40 checksum 0
01:29:14: RTP:  8012 16AF 9170 6409 0E9F 0001
01:29:14: is_rtp:1 is_frf11:0 vlen:0 delta_t:160 vofr1:0 vofr2:0
scodec:11 rtp_bits:8012 rtp_codec:18 last_bad_payload 19
01:29:14: vencap FAILED
01:29:14: PROCESS SWITCH
01:29:15: %SYS-5-CONFIG_I: Configured from console by console
01:29:34: ***SkinnyPktIp DN 3 10.2.1.1 to 40.0.0.1 pkts 4880 FAST sw
01:29:34: from_skinny_duration 150910
01:29:34: nw 3BBC2A8 addr 3BBC2A4 mac 3BBC2A4 dg 3BBC2C4 dgs 2A
01:29:34: MAC:  1841 0800
01:29:34: IP:   45B8 0046 682E 0000 3E11 E0BD 0A02 0101 2800 0001
01:29:34: TTL 62 TOS B8 prec 5
01:29:34: UDP:  6266 07D0 0032 0000
01:29:34: sport 25190 dport 2000 length 50 checksum 0
01:29:34: RTP:  8012 55FF 0057 8870 3AF4 C394
01:29:34: RTP: rtp_bits 8012 seq 55FF ts 578870 ssrc 3AF4C394
01:29:34: PAYLOAD:
01:29:34:       1409 37C9 54DE 449C 3B42 0446 3AAB 182E
01:29:34:       56BC 5184 58E5 56D3 13BE 44A7 B8C4
01:29:34:
01:29:37: ***ph_xmit_ephone DN 3 tx_pkts 6790 dest=10.2.1.1 orig len=32
 pakcopy=0 discards 31 ip_enctype 0 0 last discard: unsupported payload type
01:29:37: to_skinny_duration 153870 offset -150 last -40 seq 0 adj 0
01:29:37: IP:   45B8 003C 0875 0000 3F11 3F81 2800 0001 0A02 0101
01:29:37: TTL 63 TOS B8 prec 5
01:29:37: UDP:  07D0 6266 0028 0000
01:29:37: sport 2000 dport 25190 length 40 checksum 0
01:29:37: RTP:  8012 1AAF 9173 4769 0E9F 0001
01:29:37: is_rtp:1 is_frf11:0 vlen:0 delta_t:160 vofr1:0 vofr2:0
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone qov

To display quality of voice (QOV) statistics for calls when preset limits are exceeded, use the **debug ephone qov** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone qov [**mac-address** *mac-address*]
no debug ephone qov [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(15)ZJ2	This command was introduced for Cisco CallManager Express 3.0 and Cisco Survivable Remote Site Telephony (SRST) Version 3.0.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Once enabled, the **debug ephone qov** command produces output only when the QOV statistics reported by phones exceed preset limits. Phones are polled every few seconds for QOV statistics on VoIP calls only, not on local PSTN calls. An output report is produced when limits are surpassed for either or both of the following:

- Lost packets—A report is triggered when two adjacent QOV samples show an increase of four or more lost packets between samples. The report is triggered by an increase of lost packets in a short period of time, not by the total number of lost packets.
- Jitter and latency—A report is triggered when either jitter or latency exceeds 100 milliseconds.

To receive a QOV report at the end of each call regardless of whether the QOV limits have been exceeded, enable the **debug ephone alarm** command in addition to the **debug ephone qov** command.

The **debug ephone statistics** command displays the raw statistics that are polled from phones and used to generate QOV reports.

Examples

The following sample output describes QOV statistics for a call on ephone 5:

```
Router# debug ephone qov
Mar  7 00:54:57.329:ephone-5[7]:QOV DN 14 chan 1 (1514) ref 4 called=1514 calling=8101
Mar  7 00:54:57.329:ephone-5[7][SEP000DBDBEF37D]:Lost 91 Jitter 0 Latency 0
Mar  7 00:54:57.329:ephone-5[7][SEP000DBDBEF37D]:previous Lost 0 Jitter 0 Latency 0
Mar  7 00:54:57.329:ephone-5[7][SEP000DBDBEF37D]:Router sent 1153 pkts, current phone got
1141
received by all (shared) phones 0
Mar  7 00:54:57.329:ephone-5[7]:worst jitter 0 worst latency 0
Mar  7 00:54:57.329:ephone-5[7]:Current phone sent 1233 packets
Mar  7 00:54:57.329:ephone-5[7]:Signal Level to phone 3408 (-15 dB) peak 3516 (-15 dB)
```

The below table describes the significant fields shown in the display.

Table 5: debug ephone qov Field Descriptions

Field	Description
Lost	Number of lost packets reported by the IP phone.
Jitter, Latency	The most recent jitter and latency parameters reported by the IP phone.
previous Lost, Jitter, Latency	Values from the previous QOV statistics report that were used as the comparison points against which the current statistics triggered generation of the current report.
Router sent pkts	Number of packets sent by the router to the IP phone. This number is the total for the entire call, even if the call is moved from one phone to another during a call, which can happen with shared lines.
current phone got	Number of packets received by the phone currently terminating the call. This number is the total for the entire call, even if the call is moved from one phone to another during a call, which can happen with shared lines.
worst jitter, worst latency	Highest value reported by the phone during the call.
Current phone sent packets	Number of packets that the current phone claims it sent during the call.
Signal Level to phone	Signal level seen in G.711 voice packet data prior to the sending of the most recent voice packet to the phone. The first number is the raw sample value, converted from G.711 to 16-bit linear format and left-justified. The number in parentheses is the value in decibels (dB), assuming that 32,767 is about +3 dB. **Note** This value is meaningful only if the call uses a G.711 codec.

Related Commands

Command	Description
debug ephone alarm	Displays alarm messages for IP phones.
debug ephone statistics	Displays call statistics for IP phones.

debug ephone raw

To provide raw low-level protocol debugging display for all Skinny Client Control Protocol (SCCP) messages, use the **debug ephone raw** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone raw [**mac-address** *mac-address*]
no debug ephone raw [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone raw** command provides raw low-level protocol debug display for all SCCP messages. The debug display provides byte level display of Skinny TCP socket messages.

If the **mac-address** keyword is not used, the debug ephone raw command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of raw protocol debugging for the Cisco IP phone with MAC address 0030.94c3.E1A8:

```
Router# debug ephone raw mac-address 0030.94c3.E1A8
EPHONE raw protocol debugging is enabled for phone 0030.94C3.E1A8
1d05h: skinny socket received 4 bytes on socket [1]
0  0  0  0
1d05h:
1d05h: SkinnyMessageID = 0
```

```
1d05h: skinny send 4 bytes
4  0  0  0  0  0  0  0  0  1  0  0
1d05h: socket [1] sent 12 bytes OK (incl hdr) for ephone-(1)
1d06h: skinny socket received 4 bytes on socket [1]
0  0  0  0
1d06h:
1d06h: SkinnyMessageID = 0
1d06h: skinny send 4 bytes
4  0  0  0  0  0  0  0  0  1  0  0
1d06h: socket [1] sent 12 bytes OK (incl hdr) for ephone-(1)
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone register

To set registration debugging for the Cisco IP phone, use the **debug ephone register** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone register [**mac-address** *mac-address*]
no debug ephone register [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone register** command sets registration debugging for the Cisco IP phones.

If the **mac-address** keyword is not used, the debug ephone register command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of registration debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone register mac-address 0030.94c3.8724
Ephone registration debugging is enabled
1d06h: New Skinny socket accepted [1] (2 active)
1d06h: sin_family 2, sin_port 50778, in_addr 10.1.0.21
1d06h: skinny_add_socket 1 10.1.0.21 50778
1d06h: ephone-(1)[1] StationRegisterMessage (2/3/12) from 10.1.0.21
1d06h: ephone-(1)[1] Register StationIdentifier DeviceName SEP003094C3E1A8
```

```
1d06h: ephone-(1)[1] StationIdentifier Instance 1    deviceType 7
1d06h: ephone-1[-1]:stationIpAddr 10.1.0.21
1d06h: ephone-1[-1]:maxStreams 0
1d06h: ephone-(1) Allow any Skinny Server IP address 10.1.0.6
.
.
.
1d06h: ephone-1[1]:RegisterAck sent to ephone 1: keepalive period 30
.
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone sccp-state

To set debugging for the SCCP call state, use the **debug ephone sccp-state** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone sccp-state [**mac-address** *mac-address*]
no debug ephone sccp-state [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a phone.

Command Default

Debugging is not enabled for SCCP state.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CallManager Express (Cisco Unified CME).

This command outputs only the debug messages that correspond to SCCP messages sent to IP phones to indicate the SCCP phone call state, such as RingIn, OffHook, Connected, and OnHook. These debug messages are also included in the output for the **debug ephone detail** command among other information.

Examples

The following example sets SCCP state debugging for one Cisco Unified CME phone with the MAC address of 678B.AEF9.DAB5.

```
Router# debug ephone sccp-state mac-address 678B.AEF9.DAB5
EPHONE SCCP state message debugging is enabled
  for ephones 000B.BEF9.DFB5
*Mar  8 06:38:45.863: %ISDN-6-CONNECT: Interface Serial2/0/0:22 is now connected to 4085254871
 unknown
*Mar  8 06:38:50.487: ephone-2[13]:SetCallState line 4 DN 60(60) chan 1 ref 100 TsRingIn
*Mar  8 06:38:52.399: ephone-2[13]:SetCallState line 4 DN 60(-1) chan 1 ref 100 TsOffHook
*Mar  8 06:38:52.399: ephone-2[13]:SetCallState line 4 DN 60(-1) chan 1 ref 100 TsConnected

*Mar  8 06:38:58.415: %ISDN-6-CONNECT: Interface Serial2/0/0:22 is now connected to 4085254871
 unknown
*Mar  8 06:38:59.963: ephone-2[13]:SetCallState line 4 DN 60(-1) chan 1 ref 100 TsOnHook
*Mar  8 06:38:59.975: %ISDN-6-DISCONNECT: Interface Serial2/0/0:22 disconnected from
4085254871 , call lasted 7 seconds
```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for one or all Cisco Unified IP phones.

debug ephone shared-line-mixed

To display debugging information about mixed shared lines, use the **debug ephone shared-line-mixed** command in privileged EXEC mode. To disable debugging messages, use the **no** form of this command.

[**no**] **debug ephone shared-line-mixed** {**all** | **errors** | **events** | **info**}

Syntax Description

all	Displays all mixed shared-line debugging messages.
errors	Displays mixed shared-line error messages.
events	Displays mixed shared-line event messages.
info	Displays general information about mixed shared lines.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **debug ephone shared-line-mixed** command to show the debugging messages for Cisco Unified SCCP IP phone users in the SCCP layer of a mixed shared line.

Examples

The following is a sample output from the **debug ephone shared-line-mixed** command for an outgoing call:

```
Router# debug ephone shared-line-mixed
Mar  9 20:16:37.571: skinny_notify_shrl_state_change: shrl event 1 sccp_id 0 peer_tag 20014
 callid 53 incoming 0
Mar  9 20:16:37.571: skinny_shrl_get_call_state: dn 14, chan 1 call state 0
Mar  9 20:16:37.571: skinny_shrl_reserve_idle_chan: reserve dn 14, chan 1
Mar  9 20:16:37.571: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 1
Mar  9 20:16:37.583: skinny_process_shrl_event: event type 1 callid 53 dn 14 chan 1
Mar  9 20:16:37.583: skinny_process_shrl_callproc: dn 14, chan 1, callid 53
Mar  9 20:16:37.583: skinny_update_shrl_call_state: dn 14, chan 1, call state 13
Router#
Router#
Mar  9 20:16:45.151: skinny_notify_shrl_state_change: shrl event 2 sccp_id 112 peer_tag
20014 callid 53 incoming 0
Mar  9 20:16:45.151: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 2
Mar  9 20:16:45.155: skinny_process_shrl_event: event type 2 callid 53 dn 14 chan 1
Mar  9 20:16:45.155: skinny_update_shrl_remote: incoming 0, remote_number 2509, remote_name
 2509
Router#
Router#
Mar  9 20:16:57.775: skinny_notify_shrl_state_change: shrl event 3 sccp_id 112 peer_tag
20014 callid 53 incoming 0
Mar  9 20:16:57.779: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 3
Mar  9 20:16:57.779: skinny_process_shrl_event: event type 4 callid 53 dn 14 chan 1
Mar  9 20:16:57.779: skinny_update_shrl_call_state: dn 14, chan 1, call state 2
```

The following is a sample output from the **debug ephone shared-line-mixed** command for an incoming call with hold and resume:

```
Router# debug ephone shared-line-mixed
Mar  9 20:17:16.943: skinny_update_shrl_dn_chan: dn 14, chan 1
Mar  9 20:17:19.143: skinny_notify_shrl_state_change: shrl event 2 sccp_id 112 peer_tag
20014 callid 57 incoming 1
Mar  9 20:17:19.143: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 2
Mar  9 20:17:19.147: skinny_process_shrl_event: event type 2 callid 57 dn 14 chan 1
Mar  9 20:17:19.147: skinny_update_shrl_remote: incoming 1, remote_number 2509, remote_name
 2509
Mar  9 20:17:19.155: skinny_shrl_get_call_state: dn 14, chan 1 call state 2
Mar  9 20:17:19.155: skinny_set_shrl_remote_connect: dn 14, chan 1
Mar  9 20:17:19.159: skinny_process_shrl_event: event type 3 callid 0 dn 14 chan 1
Mar  9 20:17:19.159: skinny_update_shrl_call_state: dn 14, chan 1, call state 13
Router#
Mar  9 20:17:24.347: skinny_notify_shrl_state_change: shrl event 4 sccp_id 112 peer_tag
20014 callid 57 incoming 0
Mar  9 20:17:24.347: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 4
Mar  9 20:17:24.347: skinny_process_shrl_event: event type 5 callid 57 dn 14 chan 1
Mar  9 20:17:24.347: skinny_update_shrl_call_state: dn 14, chan 1, call state 8
Mar  9 20:17:28.307: skinny_shrl_resume_non_active_line: ref 5 line 4
Mar  9 20:17:28.307: skinny_update_shrl_call_state: dn 14, chan 1, call state 2
Mar  9 20:17:28.319: skinny_shrl_resume_non_active_line: fake redial to 2509
Mar  9 20:17:29.127: skinny_shrl_check_remote_resume: resume callid 62 holder callid 57
Mar  9 20:17:29.127: skinny_shrl_check_remote_resume: resume callid 62 holder callid 57
Mar  9 20:17:29.127: skinny_shrl_get_privacy: dn 14, chan 1 phone 2 privacy 0
Mar  9 20:17:29.135: skinny_notify_shrl_state_change: shrl event 3 sccp_id 112 peer_tag
20014 callid 57 incoming 0
Mar  9 20:17:29.135: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 3
Mar  9 20:17:29.135: skinny_shrl_set_resume_info: dn 14, chan 1
Mar  9 20:17:29.135: skinny_update_shrl_dn_chan: dn 14, chan 1
Mar  9 20:17:29.155: skinny_process_shrl_event: event type 4 callid 57 dn 14 chan 1
Router
Mar  9 20:17:42.407: skinny_notify_shrl_hold_or_resume_request: dn 14, chan 1, hold 1
Mar  9 20:17:42.411: skinny_shrl_get_privacy: dn 14, chan 1 phone 2 privacy 0
Router#
Mar  9 20:17:46.979: skinny_notify_shrl_state_change: shrl event 1 sccp_id 112 peer_tag
20014 callid 64 incoming 0
Mar  9 20:17:46.979: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 1
Mar  9 20:17:46.983: skinny_shrl_get_privacy: dn 14, chan 1 phone 2 privacy 0
Mar  9 20:17:46.987: skinny_notify_shrl_state_change: shrl event 2 sccp_id 112 peer_tag
20014 callid 64 incoming 0
Mar  9 20:17:46.987: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 2
Mar  9 20:17:46.987: skinny_process_shrl_event: event type 1 callid 64 dn 14 chan 1
Mar  9 20:17:46.987: skinny_process_shrl_event: event type 2 callid 64 dn 14 chan 1
Mar  9 20:17:46.999: skinny_set_shrl_remote_connect: dn 14, chan 1
Mar  9 20:17:46.999: skinny_set_shrl_remote_connect: dn 14, chan 1
Mar  9 20:17:47.007: skinny_process_shrl_event: event type 3 callid 0 dn 14 chan 1
Mar  9 20:17:47.007: skinny_update_shrl_call_state: dn 14, chan 1, call state 13
Mar  9 20:17:47.007: skinny_process_shrl_event: event type 3 callid 0 dn 14 chan 1
Router#
Mar  9 20:17:53.795: skinny_notify_shrl_state_change: shrl event 3 sccp_id 112 peer_tag
20014 callid 64 incoming 0
Mar  9 20:17:53.795: skinny_notify_shrl_state_change: dn = 14, chan = 1 event = 3
Mar  9 20:17:53.795: skinny_process_shrl_event: event type 4 callid 64 dn 14 chan 1
Mar  9 20:17:53.795: skinny_update_shrl_call_state: dn 14, chan 1, call state 2
```

Related Commands

Command	Description
shared-line	Creates a directory number to be shared by multiple Cisco Unified SIP IP phones.

Command	Description
shared-line sip	Adds an ephone-dn as a member of a shared directory number in the database of the Shared-Line Service Module for a mixed shared line between Cisco Unified SIP IP phones and Cisco Unified SCCP IP phones.
show shared-line	Displays information about active calls using SIP shared lines.

debug ephone state

To set state debugging for the Cisco IP phone, use the **debug ephone state** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone state [**mac-address** *mac-address*]
no debug ephone state [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on Cisco 1760 routers.

Usage Guidelines

The **debug ephone state** command sets state debugging for the Cisco IP phones.

If the **mac-address** keyword is not used, the debug ephone state command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of state debugging for the Cisco IP phone with MAC address 0030.94c3.E1A8:

```
Router# debug ephone state mac-address 0030.94c3.E1A8
EPHONE state debugging is enabled for phone 0030.94C3.E1A8
1d06h: ephone-1[1]:OFFHOOK
1d06h: ephone-1[1]:SIEZE on activeline 0
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsOffHook
1d06h: ephone-1[1]:Skinny-to-Skinny call DN 1 to DN 2 instance 1
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsRingOut
```

```
1d06h: ephone-1[1]:Call Info DN 1 line 1 ref 158 called 5002 calling 5001
1d06h: ephone-1[1]: Jane calling
1d06h: ephone-1[1]: Jill
1d06h: ephone-1[1]:SetCallState line 3 DN 2 TsRingIn
1d06h: ephone-1[1]:Call Info DN 2 line 3 ref 159 called 5002 calling 5001
1d06h: ephone-1[1]: Jane calling
1d06h: ephone-1[1]: Jill
1d06h: ephone-1[1]:SetCallState line 3 DN 2 TsCallRemoteMultiline
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsConnected
1d06h: ephone-1[1]:OpenReceive DN 1 codec 4:G711Ulaw64k  duration 10 ms bytes 80
1d06h: ephone-1[1]:OpenReceiveChannelAck 1.2.172.21 port=24010
1d06h: ephone-1[1]:StartMedia 1.2.172.22 port=24612
1d06h: DN 1 codec 4:G711Ulaw64k duration 10 ms bytes 80
1d06h: ephone-1[1]:CloseReceive
1d06h: ephone-1[1]:StopMedia
1d06h: ephone-1[1]:SetCallState line 3 DN 2 TsOnHook
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsOnHook
1d06h: ephone-1[1]:SpeakerPhoneOnHook
1d06h: ephone-1[1]:ONHOOK
1d06h: ephone-1[1]:SpeakerPhoneOnHook
1d06h: SkinnyReportDnState DN 1 ONHOOK
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone statistics

To set call statistics debugging for the Cisco IP phone, use the **debug ephone statistics** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone statistics [**mac-address** *mac-address*]
no debug ephone statistics [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone statistics** command provides a debug monitor display of the periodic messages from the Cisco IP phone to the router. These include transmit-and-receive packet counts and an estimate of drop packets. The call statistics can also be displayed for live calls using the **show ephone** command.

If the **mac-address** keyword is not used, the debug ephone statistics command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of statistics debugging for the Cisco IP phone with MAC address 0030.94C3.E1A8:

```
Router# debug ephone statistics mac-address 0030.94C3.E1A8
EPHONE statistics debugging is enabled for phone 0030.94C3.E1A8
1d06h: Clear Call Stats for DN 1 call ref 162
1d06h: Clear Call Stats for DN 1 call ref 162
```

```
1d06h: Clear Call Stats for DN 1 call ref 162
1d06h: Clear Call Stats for DN 2 call ref 163
1d06h: ephone-1[1]:GetCallStats line 1 ref 162 DN 1: 5001
1d06h: ephone-1[1]:Call Stats for line 1 DN 1 5001 ref 162
1d06h: ephone-1[1]:TX Pkts 0 bytes 0 RX Pkts 0 bytes 0
1d06h: ephone-1[1]:Pkts lost 4504384 jitter 0 latency 0
1d06h: ephone-1[1]:Src 0.0.0.0 0 Dst 0.0.0.0 0 bytes 80 vad 0 G711Ulaw64k
1d06h: ephone-1[1]:GetCallStats line 1 ref 162 DN 1: 5001
1d06h: STATS: DN 1 Packets Sent 0
1d06h: STATS: DN 2 Packets Sent 0
1d06h: ephone-1[1]:Call Stats found DN -1 from Call Ref 162
1d06h: ephone-1[1]:Call Stats for line 0 DN -1 5001 ref 162
1d06h: ephone-1[1]:TX Pkts 275 bytes 25300 RX Pkts 275 bytes 25300
1d06h: ephone-1[1]:Pkts lost 0 jitter 0 latency 0
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone video

To set video debugging for ephones, use the **debug ephone video** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone video
no debug ephone video

Syntax Description

This command has no arguments or keywords.

Command Default

Debugging is disabled for ephone video.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **debug ephone video** command sets ephone video traces, which provide information about different video states for the call, including video capabilities selection, start, and stop.

The debug ephone command debugs all ephones that are registered to the Cisco Unified CallManager Express (Cisco Unified CME) system.

You can enable or disable debugging on any number of ephones. To see the ephones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a ephone, the debug output is displayed for the directory numbers associated with the ephone.

Examples

The following is sample output for the **debug ephone video** command for ephones:

```
Router# debug ephone video
*Mar 13 16:10:02.703: SkinnyVideoCodecMatch_Caps2Caps: match capability: tx_idxcap = 4,
tx_idxpref = 3,
*Mar 13 16:10:02.703:                 rx_idxcap = 0, rx_idxpref = 0, videoBitRate = 7040
tx_mpi = 1
*Mar 13 16:10:04.711: ephone-19[1][SEPFFFA00000019]:checkToOpenMultiMedia: dn=19, chan=1
*Mar 13 16:10:04.711: ephone-19[1]:skinnyDP[19].s2s = 0
*Mar 13 16:10:04.711: ephone-19[1]:s2s is not set - hence not video capable
*Mar 13 16:10:04.719: ephone-19[1][SEPFFFA00000019]:SkinnyStartMultiMediaTransmission: chan
 1 dn 19
*Mar 13 16:10:04.723: ephone-19[1]:Accept OLC and open multimedia channel
*Mar 13 16:10:04.723: ephone-19[1][SEPFFFA00000019]:SkinnyOpenMultiMediaReceiveChannel: dn
 19 chan 1
*Mar 13 16:10:04.967: ephone-19[1][SEPFFFA00000019]:fStationOpenReceiveChannelAckMessage:
MEDIA_DN 19 MEDIA_CHAN 1
*Mar 13 16:10:04.967: ephone-19[1]:fStationOpenMultiMediaReceiveChannelAckMessage:
*Mar 13 16:10:04.967: ephone-19[1]:Other_dn == -1
sk3745-2#
*Mar 13 16:10:14.787: ephone-19[1]:SkinnyStopMedia: Stop Multimedia
*Mar 13 16:10:14.787: ephone-19[1][SEPFFFA00000019]:SkinnyCloseMultiMediaReceiveChannel:
passThruPartyID = 0, callReference = 23
*Mar 13 16:10:14.787: ephone-19[1]:SkinnyStopMultiMediaTransmission: line 1 chan 1 dn 19
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the ephone.
debug ephone detail	Sets detail debugging for the ephone.
debug ephone error	Sets error debugging for the ephone.
debug ephone message	Sets message debugging for the ephone.
debug ephone mwi	Sets MWI debugging for the ephone.
debug ephone pak	Provides voice packet level debugging and displays the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the ephone.
debug ephone state	Sets state debugging for the ephone.
debug ephone statistics	Sets statistics debugging for the ephone.
show debugging	Displays information about the types of debugging that are enabled for your router.
show ephone	Displays information about registered ephones.

debug ephone vm-integration

To display pattern manipulation information used for integration with voice-mail applications, use the **debug ephone vm-integration** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone vm-integration [**mac-address** *mac-address*]
no debug ephone vm-integration [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(7)T	This command was introduced.

Usage Guidelines

This command displays the voice-mail integration patterns that were created using the **pattern** commands in vm-integration configuration mode. The patterns are used to forward calls to a voice-mail number that is set with the **voicemail** command.

If you do not specify the **mac-address** keyword, the **debug ephone vm-integration** command debugs all Cisco IP phones that are registered to the router. To remove debugging for Cisco IP phones, enter the **no** form of this command with the **mac-address** keyword.

Examples

The following sample output shows information for the vm-integration tokens that have been defined:

```
Router# debug ephone vm-integration
*Jul 23 15:38:03.294:ephone-3[3]:StimulusMessage 15 (1) From ephone 2
*Jul 23 15:38:03.294:ephone-3[3]:Voicemail access number pattern check
*Jul 23 15:38:03.294:SkinnyGetCallState for DN 3 chan 1 IDLE
*Jul 23 15:38:03.294:called DN -1 chan 1, calling DN -1 chan 1 phone -1 s2s:0
*Jul 23 15:38:03.294:dn number for dn 3 is 19003
*Jul 23 15:38:03.294:Updated number for token 1 is 19003
*Jul 23 15:38:03.294:CDN number for dn 3 is
*Jul 23 15:38:03.294:Updated number for token 2 is
*Jul 23 15:38:03.294:Updated number for token 0 is
*Jul 23 15:38:03.294:Update is 219003*
*Jul 23 15:38:03.294:New Voicemail number is 19101219003*
```

The below table describes the significant fields shown in the display.

Table 6: debug ephone vm-integration Field Descriptions

Field	Description
token 0	First token that was defined in the pattern.
token 1	Second token that was defined in the pattern.
token 2	Third token that was defined in the pattern.

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when a user presses the Messages button on a phone.
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension reaches a busy extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and analog voice-mail systems.
voicemail	Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.

debug ephone whisper-intercom

To display debugging messages for the Whisper Intercom feature, use the **debug ephone whisper-intercom** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone whisper-intercom
no debug ephone whisper-intercom

Syntax Description

This command has no arguments or keywords.

Command Default

Debugging for Whisper Intercom is disabled.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(22)YB	This command was introduced.
12.4(24)T	This command was integrated into Cisco IOS Release 12.4(24

Usage Guidelines

This command displays debugging information about the Whisper Intercom feature configured on a directory number of a SCCP phone.

Examples

The following example displays output from the **debug ephone whisper-intercom** command:

```
Router# debug ephone whisper-intercom

ephone-1[0] Mac:1111.C1C1.0001 TCP socket:[8] activeLine:0 whisperLine:2 REGISTERED in SCCP
 ver 12/12 max_streams=3
mediaActive:0 whisper_mediaActive:0 startMedia:1 offhook:1 ringing:0 reset:0 reset_sent:0
paging 0 debug:0 caps:5
IP:10.6.2.185 9237 7970  keepalive 16 max_line 8
button 1: dn 1  number 2001 CH1   IDLE         CH2   IDLE
button 2: dn 161 number 6001  auto dial 6002 CH1   WHISPER
Preferred Codec: g711ulaw
Active Call on DN 161 chan 1 :6001 0.0.0.0 0 to 10.6.2.185 9280 via 10.6.2.185
G711Ulaw64k  160 bytes no vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn 162


ephone-2[1] Mac:1111.C1C1.0002 TCP socket:[7] activeLine:0 whisperLine:2 REGISTERED in SCCP
 ver 12/12 max_streams=3
mediaActive:0 whisper_mediaActive:1 startMedia:0 offhook:1 ringing:0 reset:0 reset_sent:0
paging 0 debug:0 caps:5
IP:10.6.2.185 9240 7970  keepalive 16 max_line 8
button 1: dn 2  number 2002 CH1   IDLE         CH2   IDLE
button 2: dn 162 number 6002  auto dial 6001 CH1   WHISPER
Preferred Codec: g711ulaw
Active Call on DN 162 chan 1 :6002 10.6.2.185 9280 to 10.6.2.254 2000 via 10.6.2.185
G711Ulaw64k  160 bytes no vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn 161 calledDn -1
```

Related Commands

Command	Description
show ephone-dn whisper	Displays information about whisper intercom ephone-dns that have been created in Cisco Unified CME.
whisper-intercom	Enables the Whisper Intercom feature on a directory number.

debug mwi relay errors

To debug message waiting indication (MWI) relay errors, use the **debug mwi relay errors** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug mwi relay errors
no debug mwi relay errors

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug mwi relay errors** command provides a debug monitor display of any error messages, when MWI Relay Server (Cisco IOS Telephony Server) is trying to do MWI Relay to extensions on remote Cisco IOS Telephony Service (ITS).

Examples

The following examples show errors when MWI Relay Server tries to do an MWI Relay to extension 7004, but location of 7004 is not known to the MWI Relay Server:

```
Router#
debug mwi relay errors

mwi-relay error info debugging is on
01:46:48: MWI-APP: mwi_notify_status: No ClientID (7004) registered
```

Related Commands

Command	Description
debug ephone mwi	Sets MWI debugging for the Cisco IOS Telephony Service router.
debug mwi relay events	Sets MWI relay events debugging for the Cisco IOS Telephony Service router.

debug mwi relay events

To set message waiting indication (MWI) relay events debugging, use the **debug mwi relay events** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug mwi relay events
no debug mwi relay events

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug mwi relay events** command provides a debug monitor display of events, when MWI Relay Server (Cisco IOS Telephony Server) is trying to do MWI Relay to extensions on remote Cisco IOS Telephony Services (ITS).

Examples

The following debugging messages are shown when the MWI Relay server tries to send MWI Information to remote client 7001 and the location of 7001 is known by the MWI Relay Server:

```
Router# debug mwi relay events

mwi-relay events info debugging is on
01:45:34: mwi_notify_status: Queued event for mwi_app_queue
01:45:34: MWI-APP: mwi_app_process_event:
01:45:34: MWI-APP: mwi_app_process_event: MWI Event for ClientID(7001)@(1.8.17.22)
```

Related Commands

Command	Description
debug ephone mwi	Sets MWI debugging for the Cisco IOS Telephony Service router.
debug mwi relay errors	Sets MWI relay errors debugging for the Cisco IOS Telephony Service router.

debug shared-line

To display debugging information about SIP shared lines, use the **debug shared-line** command in privileged EXEC mode. To disable debugging messages, use the **no** form of this command.

debug shared-line {**all** | **errors** | **events** | **info**}
no debug shared-line {**all** | **errors** | **events** | **info**}

Syntax Description

all	Displays all shared-line debugging messages.
errors	Displays shared-line error messages.
events	Displays shared-line event messages.
info	Displays general information about shared lines.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(22)YB	This command was introduced.
12.4(24)T	This command was integrated into Cisco IOS Release 12.4(24)T.

Examples

The following example shows output from the **debug shared-line all** command:

```
Router# debug shared-line all

Aug 21 21:56:56.949: //Shared-Line/EVENT/shrl_validate_newcall_outgoing:Outgoing call
validation request from AFW for  user = 20143, usrContainer = 4A7CFBDC
.Aug 21 21:56:56.949: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20143'
.Aug 21 21:56:56.949: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry not found for dn '20143'
.Aug 21 21:56:56.949: //Shared-Line/INFO/shrl_find_ccb_by_demote_dn:Demoted dn: 20143
.Aug 21 21:56:56.949: //Shared-Line/INFO/shrl_validate_newcall_outgoing:User '20143' doesn't
 exist in Shared-Line table
.Aug 21 21:56:56.957: //Shared-Line/EVENT/shrl_validate_newcall_incoming:Incominging call
validation request from AFW for user = 20141
.Aug 21 21:56:56.957: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20141'
.Aug 21 21:56:56.957: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry found [ccb = 4742EAD4]
for dn '20141'
.Aug 21 21:56:56.957: //Shared-Line/INFO/shrl_validate_newcall_incoming:User '20141' found:
 ccb = 4742EAD4, mem_count = 2
.Aug 21 21:56:56.957: //Shared-Line/EVENT/shrl_validate_newcall_incoming:Obtained call
instance inst: 0 for incoming call, incoming leg (peer_callid): 5399)
.Aug 21 21:56:56.957: //Shared-Line/INFO/shrl_update_barge_calltype:Updating shared-line
call -1 with calltype = 1
.Aug 21 21:56:56.961: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20141'
.Aug 21 21:56:56.961: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry found [ccb = 4742EAD4]
for dn '20141'
.Aug 21 21:56:56.961: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20141'
```

```
.Aug 21 21:56:56.961: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry found [ccb = 4742EAD4]
for dn '20141'
.Aug 21 21:57:01.689: %IPPHONE-6-REG_ALARM: 24: Name=SEP00141C48E126 Load=8.0(5.0)
Last=Phone-Reg-Rej
.Aug 21 21:57:04.261: //Shared-Line/EVENT/shrl_app_event_notify_handler:Event notification
 received: event = 9, callID = 5401, dn = 20141
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20141'
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry found [ccb = 4742EAD4]
for dn '20141'
.Aug 21 21:57:04.261: //Shared-Line/EVENT/shrl_process_connect:called with state = 3, callID
 = 5401, peer callID = 5399, dn = 20141, usrContainer = 4A7CACA4
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_connect_upd_callinfo:Parsed To: 20141@15.6.0.2,
 to-tag: 2ed5b927-6ad6
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_connect_upd_callinfo:Parsed Contact:
20141@15.6.0.2 for sipCallId: E8583537-6F0211DD-96A69BA1-1228BEFB@15.10.0.1
.Aug 21 21:57:04.261: //Shared-Line/EVENT/shrl_connect_upd_callinfo:Obtained call instance
 inst: 0
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_connect_upd_callinfo:CONNECT from shared line
 for incoming shared-line call.
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_find_peer_by_ipaddr:Trying to match peer for
 member 20141@15.6.0.2
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_find_peer_by_ipaddr:Matching peer [40002]
session target parsed = 15.6.0.2
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_connect_upd_callinfo:Matching member found:
20141@15.6.0.2
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_update_remote_name:Updating shared-line call
 dialog info 5401

.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_process_connect:Updated callinfo for callid:
 5401, member: '20141@15.6.0.2', peer-tag: 40002
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_process_connect:Notify remote users about
CALL-CONNECT.
.Aug 21 21:57:04.261: //Shared-Line/EVENT/shrl_send_dialog_notify:Sending NOTIFY to remote
 user:  20141@15.6.0.1
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_send_dialog_notify:Sending NOTIFY to remote
user:  20141@15.6.0.1 about state 3 on incoming call from 20141@15.6.0.2 privacy OFF
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_send_dialog_notify:Dialog msg: dir: 1, orient:
 2, local_tag: 2ed5b927-6ad6, remote_tag: 89DCF0-139B, local_uri: 20141@15.6.0.2, remote_uri:
 20143@15.10.0.1
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_send_dialog_notify:Dialog notify sent
successfully
.Aug 21 21:57:04.261: //Shared-Line/INFO/shrl_process_connect:Shared-Line '20141':
Successfully sent notify for callid: 5401
.Aug 21 21:57:04.265: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20141'
.Aug 21 21:57:04.265: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry found [ccb = 4742EAD4]
for dn '20141'
.Aug 21 21:57:04.265: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20143'
.Aug 21 21:57:04.265: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry not found for dn '20143'
.Aug 21 21:57:04.269: //Shared-Line/INFO/shrl_find_ccb_by_demote_dn:Demoted dn: 20143
.Aug 21 21:57:04.269: //Shared-Line/INFO/shrl_update_totag:Shared-Line not enabled for
'20143'
.Aug 21 21:57:04.269: //Shared-Line/EVENT/shrl_app_event_notify_handler:Event notification
 received: event = 21, callID = 5401, dn = 20141
.Aug 21 21:57:04.269: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20141'
.Aug 21 21:57:04.269: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry found [ccb = 4742EAD4]
for dn '20141'
.Aug 21 21:57:04.269: //Shared-Line/EVENT/shrl_process_callerid_update:called with state =
 7, callID = 5401, peer callID = 5399, dn = 20141
.Aug 21 21:57:04.269: //Shared-Line/INFO/shrl_process_callerid_update:Updated callinfo for
 callid: 5401, member: '20141@15.6.0.2', peer-tag: 40002
```

```
.Aug 21 21:57:04.269: //Shared-Line/EVENT/shrl_is_outbound:Check for shared line call type
 callid 5401for user = 20141
.Aug 21 21:57:04.269: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20141'
.Aug 21 21:57:04.269: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry found [ccb = 4742EAD4]
for dn '20141'
.Aug 21 21:57:04.269: //Shared-Line/EVENT/shrl_barge_type:Check for shared line call type
callid 5401for user = 20141
.Aug 21 21:57:04.269: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20141'
.Aug 21 21:57:04.269: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry found [ccb = 4742EAD4]
for dn '20141'
.Aug 21 21:57:04.273: //Shared-Line/INFO/shrl_find_ccb_by_dn:Searching Shared-Line table
for dn '20141'
.Aug 21 21:57:04.273: //Shared-Line/INFO/shrl_find_ccb_by_dn:Entry found [ccb = 4742EAD4]
for dn '20141'
.Aug 21 21:57:04.281: //Shared-Line/EVENT/shrl_notify_done_handler:NOTIFY_DONE received for
 subID: 5 respCode: 17
.Aug 21 21:57:04.281: //Shared-Line/INFO/shrl_find_ccb_by_subid:Search ccb for subid: 5
.Aug 21 21:57:04.281: //Shared-Line/INFO/shrl_find_ccb_by_subid:Found the entry ccb: 4742EAD4
 member: 20141@15.6.0.1
.Aug 21 21:57:04.281: //Shared-Line/INFO/shrl_free_spi_respinfo:Free ASNL resp info for
subID = 5
```

Related Commands

Command	Description
shared-line	Creates a directory number to be shared by multiple SIP phones.
show shared-line	Displays information about active calls using SIP shared lines.

debug voice register errors

To display debug information on voice register module errors during registration in a Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment, use the **debug voice register errors** command in privileged EXEC mode. To disable debugging, use the **no** form of the command.

debug voice register errors
no debug voice register errors

Syntax Description

This command has no arguments or keywords

Command Default

Disabled

Command Modes

Privileged EXEC mode

Command History

Cisco IOS Release	Modification
12.2(15)ZJ	This command was introduced for Cisco SIP SRST 3.0
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T for Cisco SIP SRST 3.0.
12.4(4)T	This command was added to Cisco Unified CME 3.4 and Cisco SIP SRST 3.4.

Usage Guidelines

Registration errors include failure to match pools or any internal errors that happen during registration.

Examples

Cisco Unified CME

The following is sample output for this command for a registration request with authentication enabled:

```
...
*May 6 18:07:26.971: VOICE_REG_POOL: Register request for (4901) from (10.5.49.83)
*May 6 18:07:26.971: VOICE_REG_POOL: key(9499C07A000036A3) added to nonce table
*May 6 18:07:26.975: VOICE_REG_POOL: Contact doesn't match any pools
*May 6 18:07:26.975: //4/89D7750A8005/SIP/Error/ccsip_spi_register_incoming_registration:
Registration Authorization failed with authorization header=
...
```

If there are no voice register pools configured for a particular registration request, the message “Contact doesn’t match any pools” is displayed.

When authentication is enabled and if the phone requesting registration cannot be authenticated, the message “Registration Authorization failed with authorization header” is displayed.

Cisco Unified SIP SRST

The following is sample output from this command:

```
Router# debug voice register errors
*Apr 22 11:52:54.523 PDT: VOICE_REG_POOL: Contact doesn't match any pools
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Register request for (33015) from (10.2.152.39)
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Contact doesn't match any pools.
*Apr 22 11:52:54.559 PDT: VOICE_REG_POOL: Register request for (33017) from (10.2.152.39)
*Apr 22 11:53:04.559 PDT: VOICE_REG_POOL: Maximum registration threshold for pool(3) hit
```

If there are no voice register pools configured for a particular registration request, the message "Contact doesn't match any pools" is displayed.

If the **max registrations** command is configured, when registration requests reach the maximum limit, the "Maximum registration threshold for pool (x) hit" message is displayed for the particular pool.

The below table describes the significant fields shown in the display.

Table 7: debug voice register errors Field Descriptions

Field	Description
Contact (doesn't match any pools)	Contact refers to the location of the SIP devices and the IP address.
key (*MAC address*)	Unique MAC address of a locally available individual SIP phone used to support a degree of authentication in Cisco Unified CME.
Register request for (*telephone number*) from (*IP address*).	The unique key for each registration is the telephone number.
Registration Authorization (failed with authorization header)	Registration Authorization message is displayed when **authenticate** command is configured in Cisco Unified CME.

Related Commands

Command	Description
debug voice register events	Displays debug information on voice register module events during SIP phone registrations in a Cisco Unified CME or Cisco Unified SIP SRST environment.

debug voice register events

To display debug information on voice register module events during Session Initiation Protocol (SIP) phone registrations in a Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified SIP Survivable Remote Site Telephony (SRST) environment, use the **debug voice register events** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug voice register events
no debug voice register events

Syntax Description

This command has no arguments or keywords

Command Default

Disabled

Command Modes

Privileged EXEC mode

Command History

Cisco IOS Release	Modification
12.2(15)ZJ	This command was introduced for Cisco SIP SRST 3.0
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T for Cisco SIP SRST 3.0.
12.4(4)T	This command was added to Cisco CME 3.4 and Cisco SIP SRST 3.4.

Usage Guidelines

Using the debug voice register events command should suffice to view registration activity.

Registration activity includes matching of pools, registration creation, and automatic creation of dial

peers. For more details and error conditions, you can use the debug voice register errors command.

Cisco Unified CME

The following example shows output from this command:

```
*May 6 18:07:27.223: VOICE_REG_POOL: Register request for (4901) from (1.5.49.83)
*May 6 18:07:27.223: VOICE_REG_POOL: Contact matches pool 1 number list 1
*May 6 18:07:27.223: VOICE_REG_POOL: key(4901) contact(10.5.49.83) add to contact table
*May 6 18:07:27.223: VOICE_REG_POOL: No entry for (4901) found in contact table
*May 6 18:07:27.223: VOICE_REG_POOL: key(4901) contact(10.5.49.83) added to contact
tableVOICE_REG_POOL pool->tag(1), dn->tag(1), submask(1)
*May 6 18:07:27.223: VOICE_REG_POOL: Creating param container for dial-peer 40001.
*May 6 18:07:27.223: VOICE_REG_POOL: Created dial-peer entry of type 0
*May 6 18:07:27.223: VOICE_REG_POOL: Registration successful for 4901, registration id is
2
...
```

The phone number 4901 associated with voice register pool 1, voice register dn 1, registered successfully. A dynamic normal (type 0) VoIP dial peer has been created for entry 4901. The dial peer can be verified using the **show voice register dial-peers** and **show sip-ua status registrar** commands.

Cisco Unified SIP SRST

The following is sample output from this command:

```
Router# debug voice register events
```

```
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Contact matches pool 1
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: key(91011) contact(192.168.0.2) add to contact
table
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: key(91011) exists in contact table
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: contact(192.168.0.2) exists in contact table, ref
 updated
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Created dial-peer entry of type 1
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Registration successful for 91011, registration
id is 257
```

The phone number 91011 registered successfully, and *type 1* is reported in the debug, which means that there is a preexisting VoIP dial peer.

```
Apr 22 10:50:38.119 PDT: VOICE_REG_POOL: Register request for (91021) from (192.168.0.3)
Apr 22 10:50:38.119 PDT: VOICE_REG_POOL: Contact matches pool 2
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: key(91021) contact(192.168.0.3) add to contact
table
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: key(91021) exists in contact table
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: contact(192.168.0.3) exists in contact table, ref
 updated
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: Created dial-peer entry of type 1
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: Registration successful for 91021, registration
id is 258
```

A dynamic VoIP dial peer has been created for entry 91021. The dial peer can be verified using the **show voice register dial-peers** and **show sip-ua status registrar** commands.

```
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: Register request for (95021) from (10.2.161.50)
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: key(95021) contact(10.2.161.50) add to contact
table
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: No entry for (95021) found in contact table
Apr 22 10:51:08.975 PDT: VOICE_REG_POOL: key(95021) contact(10.2.161.50) added to contact
table
Apr 22 10:51:08.979 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:08.979 PDT: VOICE_REG_POOL: Registration successful for 95021, registration
id is 259
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: Register request for (95012) from (10.2.161.50)
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: key(95012) contact(10.2.161.50) add to contact
table
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: No entry for (95012) found in contact table
Apr 22 10:51:09.023 PDT: VOICE_REG_POOL: key(95012) contact(10.2.161.50) added to contact
table
Apr 22 10:51:09.027 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:09.027 PDT: VOICE_REG_POOL: Registration successful for 95012, registration
id is 260
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: Register request for (95011) from (10.2.161.50)
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: key(95011) contact(10.2.161.50) add to contact
table
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: No entry for (95011) found in contact table
Apr 22 10:51:09.075 PDT: VOICE_REG_POOL: key(95011) contact(10.2.161.50) added to contact
table
Apr 22 10:51:09.079 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:09.079 PDT: VOICE_REG_POOL: Registration successful for 95011, registration
id is 261
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: Register request for (95500) from (10.2.161.50)
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: key(95500) contact(10.2.161.50) add to contact
table
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: No entry for (95500) found in contact table
Apr 22 10:51:09.127 PDT: VOICE_REG_POOL: key(95500) contact(10.2.161.50) added to contact
```

```
table
Apr 22 10:51:09.131 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:09.131 PDT: VOICE_REG_POOL: Registration successful for 95500, registration
id is 262
*Apr 22 11:52:54.523 PDT: VOICE_REG_POOL: Contact doesn't match any pools
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Register request for (33015) from (10.2.152.39)
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Contact doesn't match any pools
*Apr 22 11:52:54.559 PDT: VOICE_REG_POOL: Register request for (33017) from (10.2.152.39)
```

The below table describes the significant fields shown in the display.

Table 8: debug voice register events Field Descriptions

Field	Description
Contact	Indicates the location of the SIP devices and may indicate the IP address.
contact table	The table that maintains the location of the SIP devices.
key	The phone number is used as the unique key to maintain registrations of SIP devices.
multiple contact	More than one registration matches the same phone number.
no entry	The incoming registration was not found.
type 0	Normal dial peer.
type 1	Existing normal dial peer.
type 2	Proxy dial peer.
type 3	Existing proxy dial peer.
type 4	Dial-plan dial peer.
type 5	Existing dial-plan dial peer.
type 6	Alias dial peer.
type 7	Existing alias dial peer.
un-registration successful	The incoming unregister was successful.
Register request/registration id *number*	The internal unique number for each registration; useful for debugging particular registrations.

Related Commands

Command	Description
debug voice register errors	Displays debug information on voice register module errors during registration in a Cisco Unified CME or Cisco Unified SIP SRST environment.
show sip-ua status registrar	Displays all the SIP endpoints that are currently registered with the contact address.

Command	Description
show voice register dial-peers	Displays details of Cisco Unified SIP SRST configuration and of all dynamically created VoIP dial peers.

default (voice hunt-group)

To set a command to its defaults values, use the **default** command in voice hunt-group configuration mode.

default *default-value*

Syntax Description

default-value	One of the voice hunt group configuration commands. Valid choices are as follows: • hops (Peer or longest-idle voice hunt group only) • preference • timeout

Command Default

There are no default behaviors or values.

Command Modes

Voice hunt-group configuration (config-voi-hunt-group)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

Use this command to configure the default value for a voice hunt group command.

The default command instructs the voice hunt group to use the default value of the specified command whenever the hunt group is called. This has the same effect as using the no form of the specified command, but the default command clearly specifies which commands are using their default values.

To use the default values for more than one command, enter each command on a separate line.

Examples

The following example shows how to set the default values for two separate voice hunt-group commands:

```
Router(config)# voice hunt-group 4
 peer
Router(config-voi-hunt-group)# default hops
Router(config-voi-hunt-group)# default timeout
```

Related Commands

Command	Description
voice hunt-group	Defines a hunt group for SIP phones in Cisco Unified CME.

description (ephone)

To provide ephone descriptions for network management systems using an eXtensible Markup Language (XML) query, use the **description** command in ephone configuration mode. To remove a description, use the **no** form of this command.

description *string*
no application

Syntax Description

string	Allows for a maximum of 128 characters, including spaces. There are no character restrictions.

Command Default

No ephone description is configured.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

The descriptions configured with this command will appear neither on phone displays nor in show command output. Instead, they are sent to network management systems, such as CiscoView. Network management systems obtain **description** command data by sending an XML ISgetDevice request to a Cisco CME system. Cisco CME responds by sending ISDevDesc field data to the network management system, which uses the data to perform such tasks as printing descriptions on screen.

Examples

The following example provides a description for ephone 1:

```
Router(config)# ephone 1
Router(config-ephone) descri
ption S/N:SK09456FPH3, Location:SJ21- 2nd Floor E5-9, User: Smith, John
```

description (ephone-dn and ephone-dn-template)

To display a custom text-string description in the header bar of all supported Cisco Unified IP phones, use the **description** command in ephone-dn or ephone-dn-template configuration mode. To return to the default, use the **no** form of this command.

description *string*
no description

Syntax Description

string	Alphanumeric characters to be displayed in the header bar of the phone display. If spaces appear in the string, enclose the string in quotation marks. The maximum string length is 40 characters. **Note** Display behavior depends on phone firmware version.

Command Default

The extension number of the first line on the phone appears in the header bar.

Command Modes

Ephone-dn configuration (config-ephone)
Ephone-dn-template configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)T	Cisco ITS 2.0.1	This command was introduced.
12.2(11)YT	Cisco ITS 2.1	The number of characters in the string was modified.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

Use this command under the ephone-dn that is associated with the first line button on a Cisco Unified IP phone. This command is typically used to display the entire E.164 telephone number associated with the first line button in the header bar rather than just the extension number, which is the default.

This command is supported by the following IP phones:

- Cisco Unified IP Phone 7940 and 7940G
- Cisco Unified IP Phones 7960 and 7960G
- Cisco Unified IP Phone 7970
- Cisco Unified IP Pone 7971

For Cisco Unified IP Phone 7940s and 7940Gs or Cisco Unified IP Phone 7960s and 7960Gs, the *string* is truncated to 14 characters if the text string is greater than 14 characters.

For Cisco Unified IP Phone 797x, all characters in the *string* appear alternately with time and date, each for 5 seconds.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example shows how to define a header bar display for a phone on which the first line button is the extension number 50155:

```
Router(config)# ephone-dn 4

Router(config-ephone-dn)# number 50155
Router(config-ephone-dn)# description
 888-555-0155
```

The following example shows how to use an ephone-dn template to define a header bar display for a phone on which the first line button is the extension number 50155:

```
Router(config)# ephone-dn-template 3
Router(config-ephone-dn-template)# description
 “888 555-0155”
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 4

Router(config-ephone-dn)# number 50155
Router(config-ephone-dn)# ephone-dn-template 3
```

Related Commands

Command	Description
number	Configures a valid number for a Cisco Unified IP phone.

description (ephone-hunt)

To create a label for an ephone hunt group, use the **description** command in ephone-hunt configuration mode. To return this value to the default, use the **no** form of this command.

description *string*
no description

Syntax Description

string	Character string that identifies a hunt group.

Command Default

No description exists for the ephone hunt group.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

This command creates a label to identify the ephone-hunt group. This label helps make the configuration more readable.

Examples

The following example shows how to identify a hunt group for technical support agents.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 description Tech Support Hunt Group
 hops 3
 timeout 7, 10, 15
 max-timeout 25
 final 4500
```

description (voice hunt-group)

To specify a description for a voice hunt group, use the **description** command in voice hunt-group configuration mode. To remove the description, use the **no** form of this command.

description *description*
no description *description*

Syntax Description

description	Specific description of the hunt group.

Command Default

No description for the hunt group.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Examples

The following example shows how to specify a description for voice hunt-group 12 using the **description** command and presents the description in the output of the **do show run** command:

```
Router(config)# voice hunt-group 12
Router (config-voice-hunt-group)# description ?
  LINE  description for this hunt group
Router (config-voice-hunt-group)# description specific huntgroup description
Router (config-voice-hunt-group)# do show run | sec voice hunt-group
voice hunt-group 12 parallel
 timeout 0
description specific huntgroup description
```

Command	Description
voice hunt-group	Enters voice hunt-group configuration mode to create a hunt group for phones in a Cisco Unified CME system.

description (voice moh-group)

To display a brief description specific to a MOH group, use the **description** command in voice moh-group configuration mode. To remove the description, use the **no** form of this command.

description *string*
no description

Syntax Description

string	An alphanumeric string to add a brief description specific to a MOH group. Maximum length: 80 characters including spaces.

Command Default

No MOH group description is configured.

Command Modes

Voice moh-group configuration (config-voice-moh-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command allows you to type a brief text describing a specific voice-moh-group. You can use maximum 80 characters, including spaces to describe a MOH group.

Examples

The following example provides a description for voice-moh-group1:

```
Router(config)#
Router(config-voice-moh-group)#
Router(config-voice-moh-group) descri
ption this is a moh group for sales
```

Related Commands

Command	Description
voice-moh-group	Enter voice-moh-group configuration mode.
moh	Enables music on hold from a flash audio feed
multicast moh	Enables multicast of the music-on-hold audio stream.
extension-range	Specifies the extension range for a clients calling a voice-moh-group.

description (voice register pool)

To display a custom description in the header bar of Cisco IP Phone 7940 and 7940G or a Cisco IP Phone 7960 and 7960G, use the **description** command in voice register pool configuration mode. To return to the default, use the **no** form of this command.

description *string*
no description

Syntax Description

string	Allows for a maximum of 128 characters, including spaces. There are no character restrictions.

Command Default

The extension number of the first line on the phone appears in the header bar.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

Use this command to display a customized description in the header bar of a SIP phone instead of the extension number, which is the default. For example, you can display the entire E.164 telephone number associated with the first phone line.

String is truncated to 14 characters in the display of the Cisco IP Phone 7940, Cisco IP Phone 7940G, Cisco IP Phone 7960, and Cisco IP Phone 7960G.

Examples

The following example shows how to define a header bar display for a SIP phone on which the extension number is 50155:

```
Router(config)# voice register pool 4

Router(config-register-pool)# number 1 50155
Router(config-register-pool)# description
 888-555-0155
```

Related Commands

Command	Description
number (voice register pool)	Configures a valid number for a SIP phone.

description (voice register pool-type)description (voice register pool-type)

To specify the description string for a new phone model, use the **description** command in voice register pool-type mode. To remove the description string, use the **no** form of this command.

description *description*
no description *description*

Syntax Description

description string	Specifies description of the phone model.

Command Default

Description for the phone model is not defined. When the reference-pooltype command is configured, the description of the reference phone is inherited.

Command Modes

Voice Register Pool-Type Configuration (config-register-pooltype)

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to specify the description string for a new phone model. When you use the **no** form of this command, the inherited properties of the reference phone takes precedence over the default value.

Example

The following example shows how to specify the description string for a phone model using the **description** command:

```
Router(config)# voice register pool-type 9900
```

```
Router(config-register-pool-type)# description New Cisco SIP Phone 9900
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.

device-id (ephone-type)

To specify the device ID of a phone type, use the **device-id** command in ephone-type configuration mode. To reset to the default value, use the **no** form of this command.

device-id *number*
no device-id

Syntax Description

number	Device ID of the phone type. Range: 1 to 2,147,483,647. Default: 0. See the table below for a list of supported device IDs.

Command Default

Device ID is 0.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies the device ID of the type of phone being added with the ephone-type template. If this command is set to the default value of 0, the ephone-type is invalid.

Table 9: Supported Values for Ephone-Type Commands

Supported Device	device-id	device-type	num-buttons	max-presentation
Cisco Unified IP Phone 6901	547	6901	1	1
Cisco Unified IP Phone 6911	548	6911	1	10
Cisco Unified IP Phone 7915 Expansion Module with 12 buttons	227	7915	12	0 (default)
Cisco Unified IP Phone 7915 Expansion Module with 24 buttons	228	7915	24	0
Cisco Unified IP Phone 7916 Expansion Module with 12 buttons	229	7916	12	0
Cisco Unified IP Phone 7916 Expansion Module with 24 buttons	230	7916	24	0
Cisco Unified Wireless IP Phone 7925	484	7925	6	4
Cisco Unified IP Conference Station 7937G	431	7937	1	6
Nokia E61	376	E61	1	1

Examples

The following example shows the device ID is set to 376 for the Nokia E61 when creating the ephone-type template:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# device-id 376
Router(config-ephone-type)# device-name E61 Mobile Phone
```

Related Commands

Command	Description
device-name	Assigns a name to a phone type in an ephone-type template.
load	Associates a type of phone with a phone firmware file.
type	Assigns the phone type to a SCCP phone.

device-name

To assign a name to a phone type in an ephone-type template, use the **device-name** command in ephone-type configuration mode. To remove the name, use the **no** form of this command.

device-name *name*
no device-name

Syntax Description

name	String that identifies this phone type. Value is any alphanumeric string up to 32 characters.

Command Default

No name is assigned to this phone type.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies a device name for the type of phone being added with the ephone-type template.

Examples

The following example shows that the name “E61 Mobile Phone” is assigned to a phone type when creating the ephone-type template:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# device-id 376
Router(config-ephone-type)# device-name E61 Mobile Phone
```

Related Commands

Command	Description
device-id	Specifies the device ID for a phone type in an ephone-type template.

device-security-mode

To set the security mode for SCCP signaling for devices communicating with the Cisco Unified CME router globally or per ephone, use the **device-security-mode** command in telephony-service or ephone configuration mode. To return to the default, use the **no** form of this command.

device-security-mode {**authenticated** | **none** | **encrypted**}
no device-security-mode

Syntax Description

authenticated	SCCP signaling between a device and Cisco Unified CME through the secure TLS connection on TCP port 2443.
none	SCCP signaling is not secure.
encrypted	SCCP signaling between a device and Cisco Unified CME through the secure TLS connection on TCP port 2443, and the media uses Secure Real-Time Transport Protocol (SRTP).

Command Default

Device signaling is not secure.

Command Modes

Telephony-service configuration (config-telephony)
Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(15)XW	Cisco Unified CME 4.1	The **encrypted** keyword was added.
12.4(15)XY	Cisco Unified CME 4.2(1)	The **encrypted** keyword was added.
12.4(15)XZ	Cisco Unified CME 4.3	The **encrypted** keyword was added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command with Cisco Unified CME phone authentication and encryption.

Set the SCCP signaling security mode globally using this command in telephony-service configuration mode or per ephone using this command in ephone configuration mode. If you use both commands, the per-phone setting overrides the global setting.

Examples

The following example selects secure SCCP signaling for all ephones.

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authenticated
```

The following example selects secure SCCP signaling for ephone 28:

```
Router(config)# ephone 28
Router(config-ephone)# button 1:14 2:25
Router(config-ephone)# device-security-mode authenticated
```

The following example selects secure SCCP signaling for all ephones and then disables it for ephone 36:

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authentication
Router(config)# ephone 36
Router(config-ephone)# button 1:15 2:16
Router(config-ephone)# device-security-mode none
```

The following example selects encrypted secure SCCP signaling and encryption through SRTP for all ephones:

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode encrypted
```

device-type

To specify the phone type, use the **device-type** command in ephone-type configuration mode. To reset to the default value, use the **no** form of this command.

device-type *phone-type*
no device-type

Syntax Description

phone-type	Device type of the phone. See the table for a list of supported device types. Default value is the same value entered with the **ephone-type** command.

Command Default

Device type is the same value that is entered with the **ephone-type** command.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integerated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies the device type of the phone being added with the ephone-type template. The device type is set to the same value as the **ephone-type** command unless you use this command to change the value.

This command must be set to one of the following supported values.

Table 10: Supported Values for Ephone-Type Commands

Supported Device	device-id	device-type	num-buttons	max-presentation
Cisco Unified IP Phone 7915 Expansion Module with 12 buttons	227	7915	12	0 (default)
Cisco Unified IP Phone 7915 Expansion Module with 24 buttons	228	7915	24	0
Cisco Unified IP Phone 7916 Expansion Module with 12 buttons	229	7916	12	0
Cisco Unified IP Phone 7916 Expansion Module with 24 buttons	230	7916	24	0
Cisco Unified IP Conference Station 7937G	431	7937	1	6
Cisco Unified IP Phone 8941	586	8941	4	3
Cisco Unified IP Phone 8945	585	8945	4	3
Nokia E61	376	E61	1	1

Examples

The following example shows the device type set to 7915 in the ephone-type template for the Cisco Unified IP Phone 7915 Expansion Module with 12 buttons:

```
Router(config)# ephone-type 7915-12 addon
Router(config-ephone-type)# device-id 227
Router(config-ephone-type)# device-name 7915-12
Router(config-ephone-type)# device-type 7915
```

Related Commands

Command	Description
device-name	Assigns a name to a phone type in an ephone-type template.
ephone-type	Adds a Cisco Unified IP phone type by defining an ephone-type template.
load	Associates a type of phone with a phone firmware file.
type	Assigns the phone type to a SCCP phone.

dial-peer no-match isdn disconnect-cause

To disconnect the incoming ISDN call when no inbound voice dial peer is matched, use the dial-peer no-match disconnect-cause command in global configuration mode. To restore the default incoming call handling behavior, use the no form of this command.

dial-peer no-match isdn disconnect-cause cause-code
no dial-peer no-match isdn disconnect-cause cause-code

Syntax Description

cause-code	An ISDN cause code number. Range is from 1 to 188.

Command Default

Dial-peer no-match isdn disconnect-cause command is disabled. Incoming ISDN calls are not forced to disconnect if no inbound dial-peer is matched.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use this command to disconnect unathorized ISDN calls when no inbound voice or modem dial peer is matched.

Refer to the ISDN Cause Values table in the Cisco IOS Debug Command Reference, for a list of ISDN cause codes.

Examples

The following example shows that ISDN cause code 28 has been specified to match inbound voice or modem dial peers:

```
Router# dial-peer no-match disconnect-cause 28
```

Related Commands

Command	Description
show dial-peer voice	Displays configuration information for dial peers.

dialplan

To assign a dial plan to a SIP phone, use the **dialplan** command in voice register pool or voice register template configuration mode. To remove the dial plan from the phone, use the **no** form of this command.

dialplan *dialplan-tag*
no dialplan *dialplan-tag*

Syntax Description

dialplan-tag	Number that identifies the dial plan to use for this SIP phone. This is the *dialplan-tag* argument that was assigned to the dial plan with the **voice register dialplan** command. Range: 1 to 24.

Command Default

No dial plan is assigned to the phone.

Command Modes

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

You apply a dial plan to a SIP phone with this command after you create the dial plan with the **voice register dialplan** command. When the phone is reset or restarted, the dial plan file specified with this command is loaded to the phone. A phone can use only one dial plan.

A dial plan assigned to a SIP phone has priority over Key Press Markup Language (KPML), which is enabled by default on the phone.

If you use a voice register template to apply a command to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

After using the **no dialplan** command to remove a dial plan from a phone, use the **restart** command after creating a new configuration profile if the dial plan was defined with the **pattern** command. If the dial plan was defined using a custom XML file with the **filename** command, you must use the **reset** command for the change to take effect.

Examples

The following example shows that dial plan 5 is assigned to the SIP phone identified by pool 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# dialplan 5
```

The following example shows that dial plan 5 is assigned to voice register template 10:

```
Router(config)# voice register template 10
Router(config-register-temp)# dialplan 5
```

Related Commands

Command	Description
digit collect kpml	Enables KPML digit collection on a SIP phone.
filename	Specifies a custom XML file that contains the dial patterns to use for a SIP dial plan.
pattern	Defines a dial pattern for a SIP dial plan.
show voice register dialplan	Displays all configuration information for a specific SIP dial plan.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
voice register dialplan	Enters voice register dialplan configuration mode to define a dial plan for SIP phones.

dialplan-pattern

To define a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, use the **dialplan-pattern** command in telephony-service configuration mode. To disable the **dialplan-pattern** command settings, use the **no** form of this command.

dialplan-pattern *tag* *pattern* **extension-length** *extension-length* [{**extension-pattern** *extension-pattern* | **no-reg**}] [**demote**]
no dialplan-pattern *tag*

Syntax Description

tag	Identifies this dial-plan pattern. The tag is a number from 1 to 10.
pattern	Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits.
extension-length	Sets the number of extension digits that will appear as a caller ID.
extension-length	Number of extension digits. The extension length must match the length of extensions for IP phones. Range: 1 to 32.
extension-pattern	(Optional) Sets an extension number's leading digit pattern when it is different from the E.164 telephone number's leading digits as defined in the *extension-pattern* argument.
extension-pattern	(Optional) Extension number's leading digit pattern. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extension 500 to 599, and 5... would include 5000 to 5999. The length of the extension pattern must equal the value configured for the *extension-length* argument.
no-reg	(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.
demote	(Optional) Demotes the registered phone if it matches the pattern, extension-length, and extension pattern.

Command Default

No expansion pattern exists.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(11)YT	Cisco ITS 2.1	The **extension-pattern** keyword was added.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was modified. The demote keyword was added to the dialplan pattern command and the dialplan pattern tag value was increased to 1-10.

Usage Guidelines

This command creates a pattern for expanding individual abbreviated extension numbers of calling numbers into fully qualified E.164 numbers.

Use this command when configuring a network with multiple Cisco Unified CMEs to ensure that the appropriate calling number, extension or E.164 number, is provided to the target Cisco Unified CME, and appears on the phone display of the called phone. In networks that have a single Cisco Unified CME, this command is not needed.

If multiple dial-plan patterns are defined, the system matches extension numbers against the patterns in sequential order, starting with with the lowest numbered dial-plan pattern tag first. Once a pattern matches an extension number, the pattern is used to generate an expanded number. If additional patterns subsequently match the extension number, they are not used.

The **dialplan-pattern** command builds additional dial peers for the expanded numbers it creates. For example, when the ephone-dn with the number 1001 was defined, the following POTS dial peer was automatically created for it:

```
dial-peer voice 20001 pots
 destination-pattern 1001
 voice-port 50/0/2
```

When you define a dial-plan pattern that 1001 will match, such as 40855510.., a second dial peer is created so that calls to both the 1001 and 4085551001 numbers will be completed. In our example, the additional dial peer that is automatically created looks like the following:

```
dial-peer voice 20002 pots
 destination-pattern 4085551001
 voice-port 50/0/2
```

Both numbers are recognized by Cisco Unified CME as being associated with a SCCP phone.

Both dial peers can be seen with the **show telephony-service dial-peer** command.

In networks with multiple routers, you may need to use the **dialplan-pattern** command to expand extensions to E.164 numbers because local extension numbering schemes can overlap each other. Networks with multiple routers have authorities such as gatekeepers that route calls through the network. These authorities require E.164 numbers so that all numbers in the network will be unique. Use the **dialplan-pattern** command to expand extension numbers into unique E.164 numbers for registering with a gatekeeper.

Ephone-dn numbers for the Cisco IP phones must match the number in the *extension-length* argument; otherwise, the extension number cannot be expanded. For example, the following command maps all 3-digit extension numbers to the telephone number 40855501xx, so that extension 111 is expanded but the 4-digit extension 1011 is not.

```
dialplan-pattern 1 40855501.. extension-length 3
```

Using the **dialplan-pattern** command to expand extension numbers can sometimes result in the improper matching of numbers with dial peers. For example, the expanded E.164 number 2035550134 can match dial-peer destination-pattern 203, not 134, which would be the correct destination pattern for the desired extension. If it is necessary for you to use the **dialplan-pattern** command and you know that the expanded numbers might match destination patterns for other dial peers, you can manually configure the E.164 expanded number for an extension as its secondary number using the **number** command, as shown in the following example:

```
ephone-dn 23
 number 134 secondary 2035550134
```

The pattern created by the **dialplan-pattern** command is also used to enable distinctive ringing for inbound calls. If a calling-party number matches a dial-plan pattern, the call is considered an internal call and has a distinctive ring that identifies the call as internal. Any call with a calling-party number that does not match a dial-plan pattern is considered an external call and has a distinctive ring that is different from the internal ringing.

When the **extension-pattern** keyword and *extension-pattern* argument are used, the leading digits of an extension pattern are stripped and replaced with the corresponding leading digits of the dial plan. For example, the following command maps all 4xx extension numbers to the E.164 number 40855501xx, so that extension 412 corresponds to 4085550112.

```
dialplan-pattern 1 4085550100 extension-length 3 extension-pattern 4..
```

When the demote keyword is used, the dialplan-pattern command tries to demote the registered phone if it matches the pattern, extension-length, and extension-pattern.

Examples

The following example shows how to create dial-plan pattern 1 for extension numbers 5000 to 5099 with a prefix of 408555. If an inbound calling party number (4085555044) matches dial-plan pattern 1, the recipient phone will display an extension (5044) as the caller ID and use an internal ringing tone. If an outbound calling party extension number (5044) matches the same dial-plan pattern 1, the calling-party extension will be converted to an E.164 number (4085555044). The E.164 calling-party number will appear as the caller ID.

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855550.. extension-length 4 extension-pattern
 50..
```

In the following example, the **dialplan-pattern** command creates dial-plan pattern 1 for extensions 800 to 899 with the telephone prefix starting with 4085559. As each number in the extension pattern is declared with the **number** command, two POTS dial peers are created. In the example, they are 801 (an internal office number) and 4085579001 (an external number).

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855590.. extension-length 3 extension-pattern
 8..
```

The following example shows a configuration for two Cisco CME systems. One system uses 50.. and the other uses 60.. for extension numbers. Each is configured with the same two **dialplan-pattern** commands. Calls from the “50..” system to the “60..” system, and vice versa, are treated as internal calls. Calls that go across a H.323 network and calls that go to a PSTN through an ISDN interface on one of the configured Cisco CME routers are represented as E.164.

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855550.. extension-length 4 extension-pattern
 50..
```

```
Router(config-telephony)# dialplan-pattern 2 51055560.. extension-length 4 extension-pattern
 60..
```

Related Commands

Command	Description
show telephony-service dial-peer	Displays dial peer information for extensions in a Cisco CME system.

dialplan-pattern (call-manager-fallback)

To create a global prefix that can be used to expand the extension numbers of inbound and outbound calls into fully qualified E.164 numbers, use the **dialplan-pattern** command in call-manager-fallback configuration mode. To disable the **dialplan-pattern** command settings, use the **no** form of this command.

dialplan-pattern *tag pattern* **extension-length** *extension-length* [**extension-pattern** *extension-pattern*] [**no-reg**] [**demote**]
no dialplan-pattern *tag* [*pattern* **extension-length** *extension-length* **extension-pattern** *extension-pattern*] [**no-reg**] [**demote**]

Syntax Description

tag	Dial-plan string tag used before a ten-digit telephone number. The tag number is from 1 to 10.
pattern	Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits.
extension-length	Sets the number of extension digits that will appear as a caller ID.
extension-length	The number of extension digits. The extension length must match the setting for IP phones in Cisco Unified CallManager mode. The range is from 1 to 32.
extension-pattern	(Optional) Sets an extension number's leading digit pattern when it is different from the E.164 telephone number's leading digits defined in the *pattern* variable.
extension-pattern	(Optional) The extension number's leading digit pattern. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extensions 500 to 599; 5... would include extensions 5000 to 5999. The extension pattern must match the setting for IP phones in Cisco Unified CallManager mode.
no-reg	(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.
demote	(Optional) Demotes the registered phone if it matches the pattern, extension-length, and extension pattern.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco SRST 1.0	This command was introduced on the Cisco 2600 series and Cisco 3600 series multiservice routers and on the Cisco IAD2420 series.
12.2(2)XT	Cisco SRST 2.0	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.

Cisco IOS Release	Cisco Product	Modification
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(11)YT	Cisco SRST 2.1	The **extension-pattern** keyword was added.
15.1(3)T	Cisco Unified SRST 8.5	This command was modified. The demote keyword was added to the dialplan pattern command and the dialplan pattern tag value was increased to 1-10.

Usage Guidelines

The **dialplan-pattern** command builds additional dial peers. For example, if a hidden POTS dial peer is created, such as the following:

```
Router(config)# dial-peer voice 20001 pots
Router(config-dial-peer)# destination-pattern 1001
Router(config-dial-peer)# voice-port 50/0/2
```

and a dial-plan pattern is created, such as 40855510.., then an additional dial peer will be created that allows calls to both the 1001 and 4085551001 numbers. For example:

```
Router(config)# dial-peer voice 20002 pots
Router(config-dial-peer)# destination-pattern 4085551001
Router(config-dial-peer)# voice-port 50/0/2
```

Both dial peers can be seen with the **show dial-peer voice** command.

The **dialplan-pattern** command also creates a global prefix that can be used by inbound calls (calls to an IP phone in a Cisco Unified SRST system) and outbound calls (calls made from an IP phone in a Cisco Unified SRST system) to expand their extension numbers to fully qualified E.164 numbers.

For inbound calls (calls to an IP phone in a Cisco Unified SRST system) where the calling party number matches the dial-plan pattern, the call is considered a local call and has a distinctive ring that identifies the call as internal. Any calling party number that does not match the dial-plan pattern is considered an external call and has a distinctive ring that is different from the internal ringing.

For outbound calls, the **dialplan-pattern** command converts the calling party's extension number to an E.164 calling party number. Outbound calls that do not use an E.164 number and go through a PRI connection to the PSTN may be rejected by the PRI link as the calling party identifier.

If there are multiple patterns, called-party numbers are checked in numeric order, starting with pattern 1, until a match is found or until the last pattern has been checked. The valid dial-plan pattern with the lowest tag is used as a prefix to all local Cisco IP phones.

When **extension-pattern** *extension-pattern* keyword and argument are used, the leading digits of an extension pattern are stripped and replaced with the corresponding leading digits of the dial plan. For example, the following command maps all extension numbers 4xx to the PSTN number 40855501xx, so that extension 412 corresponds to 4085550112.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 4085550100 extension-length 3 extension-pattern
 4..
```

The number of *extension-pattern* argument characters must match the number set for the *extension-length* argument. For example, if the *extension-length* is 3, the *extension-pattern* can be 8.., 1.., 51., and so forth.

A dial-plan pattern is required to register the Cisco IP phone lines with a gatekeeper. The **no-reg** keyword provides the option of not registering specific numbers to the gatekeeper so that those numbers can be used for other telephony services.

When the demote keyword is used, the dialplan-pattern command tries to demote the registered phone if it matches the pattern, extension-length, and extension-pattern.

Examples

The following example shows how to create dial-plan pattern 1 for extension numbers 5000 to 5099 with a prefix of 408555. If an inbound calling party number (4085555044) matches dial-plan pattern 1, the recipient phone will display an extension (5044) as the caller ID and use an internal ringing tone. If an outbound calling party extension number (5044) matches dial-plan pattern 1, the calling party extension will be converted to an E.164 number (4085555044). The E.164 calling party number will appear as the caller ID.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 40855550.. extension-length 4 extension-pattern
 50..
```

In the following example, the **dialplan-pattern** command creates dial-plan pattern 1 for extensions 800 to 899 with the telephone prefix starting with 4085559. As each number in the extension pattern is declared with the **number** command, two POTs dial peers are created. In the example, they are 801 (an internal office number) and 4085559001 (an external number).

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 40855590.. extension-length 3 extension-pattern
 8..
```

The following example shows a configuration for two Cisco Unified SRST systems. Each is configured with the same **dialplan-pattern** commands, but one system uses 50.. and the other uses 60.. for extension numbers. Calls from the “50..” system to the “60..” system, and vice versa, are treated as internal calls. Calls that go across an H.323 network and calls that go to a PSTN through an ISDN interface on one of the configured Cisco Unified SRST routers are represented as E.164.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 40855550.. extension-length 4 extension-pattern
 50..
Router(config-cm-fallback)# dialplan-pattern 2 51055560.. extension-length 4 extension-pattern
 60..
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified SRST support and enters call-manager-fallback configuration mode.
show dial-peer voice	Displays information for voice dial peers.

dialplan-pattern (voice register)

To define a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, use the **dialplan-pattern** command in voice register global configuration mode. To disable the **dialplan-pattern** command settings, use the **no** form of this command.

dialplan-pattern *tag* *pattern* **extension-length** *extension-length* [{**extension-pattern** *extension-pattern* | **no-reg**}] [**demote**]
no **dialplan-pattern** *tag*

Syntax Description

tag	Unique number for identifying this dial-plan pattern. Range: 1 to 10.
pattern	Dial-plan pattern to be matched, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits.
extension-length	Number of extension digits that will appear as a caller ID.
extension-length	Number of digits in an extension. This variable must match the length of the directory numbers configured for SIP extensions in Cisco Unified CME. Range: 1 to 32.
extension-pattern	(Optional) Leading digit pattern to be configured for an extension when it is different from the leading digit pattern of the E.164 telephone number, as defined in the *extension-pattern* argument.
extension-pattern	(Optional) Leading digit pattern to be stripped from extension number when expanding an extension to an E.164 telephone number. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extension 500 to 599, and 5... would include 5000 to 5999. The length of the extension pattern must equal the value configured for the *extension-length* argument.
no-reg	(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.
demote	(Optional) Demotes the registered phone if it matches the pattern,e xtension-length, and extension pattern.

Command Default

No expansion pattern exists.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS 12.4(9)T.

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was modified. The demote keyword was added to the dialplan pattern command and the dialplan pattern tag value was increased to 1-10.

Usage Guidelines

This command creates a pattern for expanding individual abbreviated SIP extension numbers of calling numbers into fully qualified E.164 numbers.

Use this command when configuring a network with multiple Cisco Unified CMEs to ensure that the appropriate calling number, extension or E.164 number, is provided to the target Cisco Unified CME, and appears on the phone display of the called phone. In networks that have a single Cisco Unified CME, this command is not needed.

Up to five dial-plan patterns can be configured. If multiple dial-plan patterns are defined, the system matches extension numbers against the patterns in sequential order, starting with the lowest numbered dial-plan pattern tag first.

Dial peers for directory numbers are automatically created when SIP phones register in Cisco Unified CME. The **dialplan-pattern** command builds a second dial peer for the expanded number because an extension number matches the pattern. Both numbers are recognized by Cisco Unified CME as being associated with a SIP phone.

For example, the following POTS dial peer is automatically created for extension number 1001 when the associated SIP phone registers in Cisco Unified CME:

```
dial-peer voice 20001 pots
 destination-pattern 1001
 voice-port 50/0/2
```

If the extension number (1001) also matches a dial-plan pattern that is configured using the **dialplan-pattern** command, such as 40855510.., a second dial peer is dynamically created so that calls to both the 1001 and 4085551001 numbers can be completed. Based on the dial-plan pattern to be matched, the following additional POTS dial peer is created:

```
dial-peer voice 20002 pots
 destination-pattern 4085551001
 voice-port 50/0/2
```

Using the **no** form of this command will remove the dial peer that was created for the expanded number.

All dial peers can be displayed by using the **show dial-peer voice summary** command. All dial peers for numbers associated to SIP phones only can be displayed by using the **show voice register dial-peers** command. Dial peers created by using the **dialplan-expansion** command cannot be seen in the running configuration.

The value of the extension-length argument must be equal to the length of extension number to be matched, otherwise, the extension number cannot be expanded. For example, the following command maps all 3-digit extension numbers to the telephone number 40855501.., so that extension 111 is expanded but 4-digit extension number 1111 is not.

```
dialplan-pattern 1 40855501.. extension-length 3
```

When the **extension-pattern** keyword and *extension-pattern* argument are configured, the leading digits of the extension pattern variable are stripped away and replaced with the corresponding leading digits of the dial-plan pattern to create the expanded number. For example, the following command maps all 3-digit extension numbers with the leading digit of "4" to the telephone number 40855501.., so that extension 434 corresponds to 4085550134.

```
dialplan-pattern 1 40855501.. extension-length 3 extension-pattern 4..
```

To apply dialplan-pattern expansion on a per-system basis to individual SIP *redirecting* numbers in a Cisco Unified CME system, including original called and last reroute numbers, use the **call-forward** command.

When the demote keyword is used, the dialplan-pattern command tries to demote the registered phone if it matches the pattern, extension-length, and extension-pattern

Examples

The following example shows how to create a dialplan-pattern for expanding extension numbers 60xxx to E.164 numbers 5105555xxx.

```
Router(config)# voice register global
Router(config-register-global)# dialplan-pattern 1 5105550... extension-length 5
```

The following example is output from the **show dial-peer summary** command displaying information for four dial peers, one each for extensions 60001 and 60002 and, because the dialplan-expansion command was configured to expand 6.... to 4085555...., one each for 4085550001 and 4085550002. The latter two dial peers will not appear in the running configuration.

```
Router# show dial-peer summary
             AD                                    PRE PASS                OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN      FER THRU SESS-TARGET    STATT
20010  pots  up   up             60002$             0                         0
20011  pots  up   up             60001$             0                         9
20012  pots  up   up             5105555001$        0                         9
20013  pots  up   up             5105555002$        0                         0
```

Related Commands

Command	Description
call-forward (voice register)	Applies dial-plan pattern expansion globally to redirecting number.
show dial-peer summary	Displays all dial peers created in Cisco Unified CME.
show voice register dial-peer	Displays dial-peer information for SIP extensions in Cisco Unified CME.

digit collect kpml

To enable Key Press Markup Language (KPML) digit collection on a SIP phone, use the **digit collect kpml** command in voice register pool or voice register template configuration mode. To disable KPML, use the **no** form of this command.

digit collect kpml
no digit collect kpml

Syntax Description

This command has no arguments or keywords.

Command Default

KPML digit collection is enabled.

Command Modes

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

Command History

Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

KPML is enabled by default for all directory numbers on the phone. A dial plan assigned to a phone has priority over KPML. Use the **no digit collect kpml** command to disable KPML on a phone.

If you use a voice register template to apply a command to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

KPML is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

Examples

The following example shows KPML enabled on SIP phone 4:

```
Router(config)# voice register pool 4
Router(config-register-pool)# digit collect kpml
```

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.
show voice register pool	Displays all configuration information associated with a SIP phone.
voice register dialplan	Enters voice register dialplan configuration mode to define a dial plan for SIP phones.

direct-inward-dial isdn

To enable incoming ISDN enbloc dialing calls, use the direct-inward-dial isdn command in voice service voip mode. To disable incoming ISDN enbloc dialing calls use the no form of the command.

direct-inward-dial isdn
no direct-inward-dial isdn

Syntax Description

This command has no arguments or keywords.

Command Default

The direct inward dial isdn is command is enabled.

Command Modes

voice service pots

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use the direct-inward-dial-isdn command to enable the direct-inward-dial (DID) call treatment for an incoming ISDN call. When this feature is enabled, the incoming ISDN call is treated as if the digits were received from the DID trunk. The called number is used to select the outgoing dial peer. No dial tone is presented to the caller to collect dialed digits even if “no direct-inward-dial” of the selected inbound dial-peer is defined for an incoming ISDN call.

Use the no form of this command to turn off the global direct-inward-dial setting for incoming ISDN calls. When this command line is disabled, the “direct-inward-dial” setting of a selected inbound dial-peer is used to handle the incoming ISDN calls.'

Examples

The following is a sample output from this command displaying DID enabled for ISDN:

```
!
voice service voip
 ip address trusted list
  ipv4 172.19.245.1
  ipv4 172.19.247.1
  ipv4 172.19.243.1
  ipv4 171.19.245.1
  ipv4 171.19.10.1
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service media-renegotiate
 sip
  registrar server expires max 120 min 120
!
!
dial-peer voice 1 voip
 destination-pattern 5511...
 session protocol sipv2
 session target ipv4:1.3.45.1
 incoming called-number 5522...
 direct-inward-dial
```

```
 . . .
!
```

Related Commands

Command	Description
voice service	Enters voice service configuration mode.

directory

To define the order in which the names of Cisco IP phone users are displayed in the local directory, use the **directory** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

directory {**first-name-first** | **last-name-first**}
no directory {**first-name-first** | **last-name-first**}

Syntax Description

first-name-first	First name is entered first in the Cisco IP phone directory name field.
last-name-first	Last name is entered first in the Cisco IP phone directory name field.

Command Default

Default is **first-name-first**.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

This command defines name order in the local directory. The directory itself is generated from entries made using the **name** command and the **number** command in ephone-dn configuration mode.

Note The name information must be entered in the correct order in the **name** command.

The location for the file that is accessed when the Directories button is pressed is specified in the **url** (telephony-service) command.

Examples

The following example shows how to configure the local directory with the last name first:

```
Router(config)# telephony-service
Router(config-telephony)# directory last-name-first
```

Related Commands

Command	Description
name	Specifies a name to be associated with an extension (ephone-dn).
number	Specifies a telephone number to be associated with an extension (ephone-dn).
url	Provisions URLs for the displays associated with buttons on Cisco IP phones.

directory entry

To add a system-wide phone directory and speed-dial definition, use the **directory entry** command in telephony-service configuration mode. To remove a definition, use the **no** form of this command.

directory entry {*directory-tag number* **name** *name* | **clear**}
no directory entry {*directory-tag* | **clear**}

Syntax Description

directory-tag	Digit string that provides a unique identifier for this entry. Range: 1 to 250.
number	String of up to 32 digits that provides the full telephone number for this entry.
name *name*	String of up to 24 alphanumeric characters, including spaces. Cannot include opening or closing quotation marks (‘, ’ , “, or ”).
clear	Removes all directory entries that were made with this command.

Command Default

Entries do not exist.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)XL	Cisco CME 3.2.1	This feature was modified to enable systemwide speed-dialing of entries from 34 to 99.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(15)XZ	Cisco Unified CME 4.3	The maximum number of directory entries was increased from 100 to 250.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Cisco Unified CME automatically creates a local phone directory consisting of the telephone numbers and names that are entered during ephone-dn configuration. Additional directory entries can be made by administrators using the **directory entry** command. Phone number directory listings are displayed in the order in which they are entered.

A single entry can be removed using the **no directory entry directory-tag** command.

Directory entries that have directory-tag numbers from 34 to 99 also can be used as system-wide speed-dial numbers. That is, if you have the following definition for the headquarters office, any phone user can speed-dial the number:

```
Router(config)# telephony-service
```

```
Router(config-telephony)# directory entry 51 4085550123 name Headquarters
```

Analog phone users press the asterisk (*) key and the speed-dial identifier (tag number) to dial a speed-dial number.

IP phone users follow this procedure to dial a speed-dial number:

1. With the phone on-hook, an IP phone user presses a two-digit speed-dial code (that is, 05 for the entry with tag 5). A new soft key, Abbr, appears in the phone display.
2. The phone user picks up the phone handset and presses the Abbr soft key. The full telephone number associated with the speed-dial tag is dialed.

Examples

The following example adds six telephone listings to the local directory. The last two entries, with the identifiers 50 and 51, can be speed-dialed by anyone on the system because their identifiers (directory-tags) are between 34 and 99.

```
Router(config)# telephony-service
Router(config-telephony)# directory entry 1 4045550110 name Atlanta
Router(config-telephony)# directory entry 2 3125550120 name Chicago
Router(config-telephony)# directory entry 4 2125550140 name New York City
Router(config-telephony)# directory entry 5 2065550150 name Seattle
Router(config-telephony)# directory entry 50 4085550123 name Corp Headquarters
Router(config-telephony)# directory entry 51 4085550145 name Division Headquarters
```

Related Commands

Command	Description
show telephony-service directory-entry	Displays the configured directory entries.
url directories	Provisions the directory URL to select an external directory resource and disables the Cisco Unified CME local directory service.

display-logout

To specify a message to display on phones in an ephone hunt group when all phones in the hunt group are logged out, use the **display-logout** command in ephone-hunt configuration mode. To return this value to the default, use the **no** form of this command.

display-logout *string*
no display-logout

Syntax Description

string	Character string to be displayed on hunt group member IP phones when all members are logged out.

Command Default

No logout message exists.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

This command defines a plain-text message that displays on phones with ephone-dns that are members of a hunt group when all the members of the group are logged out. The message can be used to notify agents that no agents are available to take hunt group calls. It can also be used to tell agents about the disposition of any incoming calls to the hunt group when no agents are available to answer calls. For example, you could set the display to read “All Agents Unavailable,” or “Hunt Group Voice Mail” or “Hunt Group Night Service.”

Examples

The following example specifies a message to display when all agents are logged out of hunt group 3.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 display-logout All Agents Logged Out
 hops 3
 timeout 7, 10, 15
 max-timeout 25
 final 4500
```

dnd (voice register pool)

To enable the Do-Not-Disturb (DND) feature, use the **dnd-control** command in voice register pool configuration mode. To disable the DND, use the **no** form of this command.

dnd
no dnd

Syntax Description

This command has no arguments or keywords.

Command Default

DND is disabled

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Examples

The following example shows how to enable DND:

```
Router(config)# voice register pool 1
Router(config-register-pool)# dnd
```

Related Commands

Command	Description
dnd-control (voice register template)	Enables DND soft key in template to be assigned to SIP phones in Cisco Unified CME.

dnd feature-ring

To disable ringing on phone buttons configured for feature ring when the phone is in do-not-disturb (DND) mode, use the **dnd feature-ring** command in ephone configuration mode. To allow lines configured for feature ring to ring when the phone is in DND mode, use the **no** form of this command.

dnd feature-ring
no dnd feature-ring

Syntax Description

This command has no arguments or keywords.

Command Default

Incoming calls to buttons configured for feature ring do not ring in DND mode.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command applies only to phone lines that are configured for the feature-ring option with the **button f** command.

Note that the affirmative form of the command is enabled by default and feature-ring lines will not ring when the phone is in DND mode. To enable feature-ring lines to ring when the phone is in DND mode, use the **no dnd feature-ring** command.

Examples

For the following example, when DND is active on ephone 1 and ephone 2, button 1 will ring, but button 2 will not.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1001

Router(config)# ephone-dn 2

Router(config-ephone-dn)# number 1002

Router(config)# ephone-dn 10

Router(config-ephone)# number 1110
Router(config-ephone)# preference 0
Router(config-ephone)# no huntstop

Router(config)# ephone-dn 11
Router(config-ephone)# number 1111
Router(config-ephone)# preference 1
Router(config-ephone)# no huntstop

Router(config)# ephone 1
Router(config-ephone)# button 1f1
Router(config-ephone)# button 2o10,11
Router(config-ephone)# no dnd feature-ring
```

```
Router(config-ephone-dn)# ephone 2
Router(config-ephone)# button 1f2
Router(config-ephone)# button 2o10,11
Router(config-ephone)# no dnd feature-ring
```

Related Commands

Command	Description
button	Associates ephone-dns with individual buttons on a Cisco IP phone and specifies ring behavior.

dnd-control (voice register template)

To enable the Do-Not-Disturb (DND) soft key on SIP phones, use the **dnd-control** command in voice register template configuration mode. To disable the DND soft key on a SIP phone, use the **no** form of this command.

dnd-control
no dnd-control

Syntax Description

This command has no arguments or keywords.

Command Default

DND soft key is enabled on SIP phones in Cisco Unified CME.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command enables a soft key for Do-Not-Disturb (DND) in the specified template which can then be applied to SIP phones. The DND soft key is enabled by default. To disable the DND soft key, use the **dnd** command. To apply a template to a SIP phone, use the template command in voice register pool configuration mode.

Examples

The following example shows how to disable the DND soft key:

```
Router(config)# voice register template 1
Router(config-register-template)# dnd-control
```

Related Commands

Command	Description
dnd (voice register pool)	Enables DND feature.

dn-webedit

To enable the adding of extensions (ephone-dns) through the Cisco Unified CME graphical user interface (GUI), use the **dn-webedit** command in telephony-service configuration mode. To disable this feature, use the **no** form of this command.

dn-webedit
no dn-webedit

Syntax Description

This command has no arguments or keywords.

Command Default

Extensions cannot be added through the Cisco Unified CME GUI.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Usage Guidelines

The **dn-webedit** command enables the adding of extensions through the web-based GUI. If the **dn-webedit** command is enabled, a customer administrator or a system administrator can modify and assign extensions associated with the Cisco Unified CME router. If this ability is disabled, extensions must be added using Cisco IOS commands.

If the set of extension numbers used by the router is part of a larger telephone network, limitations on modification might be needed to ensure network integrity. Disabling the **dn-webedit** command prevents an administrator from allocating phone numbers and prevents assignment of numbers that may already be used elsewhere in the network.

Examples

The following example enables editing of directory numbers through the web-based GUI interface:

```
Router(config)# telephony-service
Router(config-telephony)# dn-webedit
```

Related Commands

Command	Description
time-webedit	Enables time setting through the web interface.

dst (voice register global)

To set the time period for daylight saving time on SIP phones, use the **dst** command in voice register global configuration mode. To disable daylight saving time, use the **no** form of this command.

dst auto-adjust
no dst {**start** | **stop**}

Syntax Description

start	Sets beginning time for daylight saving time.
stop	Sets ending time for daylight saving time.
month	Abbreviated month. The following abbreviations are valid: **jan**, **feb**, **mar**, **apr**, **may**, **jun**, **jul**, **aug**, **sep**, **oct**, **nov**, **dec**.
day *day-of-month*	Date of the month. Range is 1 to 31.
week *week-number*	Number identifying the week of the month. Range is 1 to 4, or 8, where 8 represents the last week of the month.
day *day-of-week*	Abbreviated day of the week. The following abbreviations are valid: **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, **sat**.
time *hour:minutes*	Beginning and ending time for daylight saving time, in HH:MM format using a 24-hour clock. The stop time must be greater than the start time. The value 24:00 is not valid. If you enter 00:00for both start time and stop time, daylight saving time is enabled for the entire 24-hour period on the specified date.

Command Default

Default start time is first week of April, Sunday, 2:00 a.m and default stop time is last week of October, Sunday 2:00 a.m.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command sets the stop and start times for daylight saving time if the **dst auto-adjust** command is configured.

Examples

The following example shows how to set automatic adjustment of daylight saving time:

```
Router(config)# voice register global
Router(config-register-global)# dst start Jan day 1 time 00:00
Router(config-register-global)# dst stop Mar day 31 time 23:99
```

Related Commands

Command	Description
date-format (voice register global)	Sets the date display format on SIP phones in a Cisco CME system.
dst auto-adjust (voice register global)	Enables automatic adjustment of daylight saving time on SIP phones.
time-format (voice register global)	Selects a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a SIP CME system.
timezone (voice register global)	Sets the time zone used for SIP phones in a Cisco CME system.

dst auto-adjust (voice register global)

To enable automatic adjustment of daylight saving time on SIP phones, use the **dst auto-adjust** command in voice register global configuration mode. To disable daylight saving time auto adjustment, use the **no** form of this command.

dst auto-adjust
no dst auto-adjust

Syntax Description

This command has no arguments or keywords.

Command Default

Automatic adjustment of daylight saving time on SIP phones is enabled.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

Automatic adjustment for daylight saving time is enabled by default. To disable auto adjusting for DST, use the **no dst auto-adjust** command. To set the start and stop times for DST, use the **dst** command.

Examples

The following example shows how to disable the automatic adjustment for daylight saving time:

```
Router(config)# voice register global
Router(config-register-global)# no dst auto-adjust
```

Related Commands

Command	Description
date-format (voice register global)	Sets the date display format on SIP phones in a Cisco CME system.
dst (voice register global)	Sets the start and stop time if using daylight saving time on SIP phones.
time-format (voice register global)	Selects a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a SIP CME system.
timezone (voice register global)	Sets the time zone used for SIP phones in a Cisco CME system.

dtmf-relay (voice register pool)

To specify the list of DTMF relay methods that can be used to relay dual-tone multifrequency (DTMF) audio tones between Session Initiation Protocol (SIP) endpoints, use the **dtmf-relay** command in voice register pool configuration mode. To send the DTMF audio tones as part of an audio stream, use the **no** form of this command.

dtmf-relay [**cisco-rtp**] [**rtp-nte**] [**sip-notify**] [**sip-kpml**]
no dtmf-relay

Syntax Description

cisco-rtp	Forwards DTMF audio tones by using Real-Time Transport Protocol (RTP) with a Cisco proprietary payload type. This keyword is supported only for dial peers that are created by incoming REGISTERs from a SIP gateway. It is not supported for dial peers that are created by a SIP Cisco IP phone.
rtp-nte	Forwards DTMF audio tones by using Real-Time Transport Protocol (RTP) with a Named Telephone Event (NTE) payload.
sip-notify	Forwards DTMF audio tones by using SIP-NOTIFY messages. This keyword is supported only for dial peers that are created by incoming REGISTERs from a SIP gateway. It is not supported for dial peers that are created by a SIP Cisco IP phone.
sip-kpml	Forwards DTMF audio tones through Keypad Markup Language (KPML) messages.

Command Default

DTMF tones are disabled and sent in-band. That is, they remain in the audio stream.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(4)T	Cisco SIP SRST 3.0	This command was introduced.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco Unified CME.
15.1(1)T1	Cisco Unified CME 8.1 Cisco SIP SRST 8.1	The sip-kpml keyword was added to this command.

Usage Guidelines

During Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CME registration, a dial peer is created and that dial peer has a default DTMF relay of in-band.

This command command allows you to change the default to a desired value. You must use one or more keywords when configuring this command.

DTMF audio tones are generated when you press a button on a Touch-Tone phone. The tones are compressed at one end of the call and when the digits are decompressed at the other end, there is a risk that they can become distorted. DTMF relay reliably transports the DTMF audio tones generated after call establishment out-of-band.

The SIP Notify method sends Notify messages bidirectionally between the originating and terminating gateways for a DTMF event during a call. If multiple DTMF relay mechanisms are enabled on a SIP dial peer and are negotiated successfully, the SIP Notify method takes precedence.

SIP Notify messages are advertised in an Invite message to the remote end only if the **dtmf-relay** command is set.

For SIP calls, the most appropriate methods to transport DTMF tones are RTP-NTE or SIP-NOTIFY.

Note The **cisco-rtp** keyword is a proprietary Cisco implementation. If the proprietary Cisco implementation is not supported, the DTMF relay feature does not function, and the gateway sends DTMF tones in-band.

- The **sip-notify** keyword is available only if the VoIP dial peer is configured for SIP.

Examples

Cisco Unified CME

The following example shows how to enable the RTP-NTE and SIP-NOTIFY mechanisms for DTMF relay for SIP phone 4:

```
Router(config)# voice register pool 4
Router(config-register-pool)# dtmf-relay rtp-nte sip-notify
```

The following example shows sip-kpml option configured for dtmf-relay in voice register pool 5:

```
Router#show running config
voice register global
 mode cme
 source-address 10.32.153.49 port 5060
 max-dn 200
 max-pool 100
!
voice register pool  5
 id mac 0023.3319.8B7B
 type 7945
 number 1 dn 5
 dtmf-relay sip-kpml
 username betaone password cisco
 codec g711ulaw
 no vad
```

Cisco Unified SIP SRST

The following is sample output from the **show running-config** command that shows that voice register pool 1 has been set up to send DTMF tones:

```
voice register pool 1
 application SIP.app
 incoming called-number 308
 voice-class codec 1
 dtmf-relay rtp-nte
```

Related Commands

Command	Description
dtmf-relay (voice over IP)	Specifies how an H.323 or SIP gateway relays DTMF tones between telephony interfaces and an IP network.

Cisco Unified CME Commands: E

elin

To create a PSTN number that replaces a 911 caller's extension, use the **elin** command in voice emergency response location configuration mode. To remove the number, use the **no** form of this command.

elin {**1** | **2**} *number*
no elin [{**1** | **2**}]

Syntax Description

{**1** | **2**}	Number index.
number	PSTN number that replaces a 911 caller's extension.

Command Default

No replacement number is created.

Command Modes

Voice emergency response location configuration (cfg-emrgncy-resp-location)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added for Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to specify an ELIN, a PSTN number that will replace the caller's extension.

The PSAP will see this number and use it to query the ALI database to locate the caller. The PSAP also uses this command for callbacks.

You can configure a second ELIN using the **elin 2** command. If two ELINs are configured, the system selects an ELIN using a round-robin algorithm. If an ELIN is not defined for the ERL, the PSAP sees the original calling number.

Examples

In the following example, all IP phones with the IP address of 10.X.X.X or 192.168.X.X are automatically associated with this ERL. If one of the phones dials 911, its extension is replaced with 408 555-0100 before it goes to the PSAP. The PSAP will see that the caller's number is 408 555-0100.

```
voice emergency response location 1
 elin 1 4085550100
 subnet 10.0.0.0 255.0.0.0
 subnet 2 192.168.0.0 255.255.0.0
```

Related Commands

Command	Description
subnet	Defines which IP phones are part of this ERL.

elin (voice emergency response settings)

To create a default ELIN that is used if no ERL has a subnet mask that matches the current 911 caller's IP phone address, use the **elin** command in voice emergency response settings configuration mode. To remove the number, use the **no** form of this command.

elin *number*
no elin

Syntax Description

number	An E.164 number to be used as the default ELIN.

Command Default

No default ELIN number is created.

Command Modes

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to specify an E.164 number to be the default ELIN if the 911 caller's IP phone address does not match the subnet of any location in any ERL zone. The default ELIN can be an existing ELIN already defined in an ERL or it can be unique.

Examples

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller's IP phone address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500.

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
```

Related Commands

Command	Description
callback	Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.
expiry	Number of minutes a 911 call is associated to an ELIN in case of a callback from the 911 operator.
logging	Syslog informational message printed to the console each time an emergency call is made.

Command	Description
voice emergency response settings	Creates a tag for identifying settings for E911 behavior.

em external

To remove the login page under the Extension Mobility option from the Services menu on IP phones in Cisco Unified CME, use the **em external** command in telephony-service configuration mode. To return to default, use the **no** form of this command.

em external
no em external

Syntax Description

This command has no keywords or arguments.

Command Default

Login page for Extension Mobility is accessible under the Extension Mobility option in the Services menu.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command removes the Extension Mobility login page from the Sevices menu on all IP phones registered in a Cisco Unified CME system on which Extension Mobility is enabled.

Examples

The following partial output shows the configuration for this command:

```
router# show running-configuration

.
.
.
telephony-service
 em external
 em logout 1:0
 max-ephones 10
 max-dn 100
 ip source-address 10.0.0.1 port 2000
 url authentication http://10.0.0.1/CCMCIP/authenticate.asp
 cnf-file location flash:
 cnf-file perphone
 max-conferences 8 gain -6
 transfer-system full-consult
 create cnf-files version-stamp Jan 01 2002 00:00:00
!
```

Related Commands

Command	Description
ip http server	Enables the HTTP server on the Cisco Unified CME router that hosts the service URL for the Extension Mobility Login and Logout page.

em keep-history

To disable Automatic Clear Call History for Extension Mobility phones in Cisco Unified CME, use the **em keep-history** command in telphony-service configuration mode. To return to the default, use the **no** form of this command.

em keep-history
no em keep-history

Syntax Description

This command has no arguments or keywords.

Command Default

Call history record is automatically cleared when a user logs out from an Extension Mobility phone.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command disables Automatic Clear Call History for Extension Mobility phones in Cisco Unified CME.

In Cisco Unified CME 4.3 and later versions, the EM manager in Cisco Unified CME sends commands to a phone to clear call history anytime a user is logs out from Extension Mobility. Use this command in telephony-service configuration mode to disable this feature at a system-level.

Examples

The following example shows how to configure Extension Mobility in Cisco Unified CME to keep, not clear, call histories after users log out from Extension Mobility phones:

```
Router(config)# telephony-service
Router(config-telephony)# em keep-history
Router(config-telephony)#
```

em logout

To configure up to three time-of-day based timers for automatically logging out all Extension Mobility users, use the **em logout** command in telephony-service configuration mode. To disable the timer, use the **no** form of this command.

em logout *time1* [*time2*] [*time3*]
no command *time1* [*time2*] [*time3*]

Syntax Description

time	Time of day after which all users that are logged into Extension Mobility are logged out from Extension Mobility. Range: 00:00 to 24:00 on a 24-hour clock.

Command Default

No time-of-day timer is created for automatically logging out Extension Mobility users.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command creates up to three time-of-day timers for automatically logging out all Extension Mobility users. If an Extension Mobility user is using the phone when automatic logout occurs, the user is logged out after the active call is completed.

The call history record is automatically cleared when a user logs out from an Extension Mobility phone. To disable Automatic Clear Call History on all Extension Mobility phones, use the **em keep-history** command in telephony-service configuration mode.

Examples

The following example shows how to configure two time-of-day timers to automatically log out all logged-on Extension Mobility users at 5:30 PM and again at midnight each day:

```
Router(config)# telephony-service
Router(config-telephony)# em logout 17:30 24:00
Router(config-telephony)#
```

Related Commands

Command	Description
em keep-history	Disables Automatic Clear Call History for Extension Mobility in Cisco Unified CME.

emadmin login

To permit an external application to log into a Cisco Unified IP phone that is enabled for Extension Mobility in Cisco Unified CME, use the **emadmin login** command in privileged EXEC mode.

emadmin login *name ephone-tag*

Syntax Description

name	Credential for Extension Mobility. This credential must be already configured by using the **user** command in voice-user-profile configuration mode.
ephone-tag	Unique identifier for IP phone that is enabled for Extension Mobility. This tag must already be configured by using the **ephone** command.

Command Default

External application cannot log into an Extension Mobility phone.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command enables an external application, such as a CSTA client application, to log into an Extension Mobility phone.

Before using this command, configure a credential in Extension Mobility by using the **user** command in voice-user-profile configuration mode.

The IP phone to be accessed must be enabled for Extension Mobility.

The application remains logged into the phone until it is manually or automatically logged out from the Extension Mobility phone

This command does not have a **no** form.

Examples

The following example shows how to configure this command to log an application into an Extension Mobility phone (2) using the “user204” credential:

```
Router# login user204
 2
Router#
```

Related Commands

Command	Description
emadmin logout	Logs out an external application from Extension Mobility.
em logout	Creates up to three time-of-day timers for automatically logging out all Extension Mobility users.

Command	Description
logout-profile	Enables an IP phone for Extension Mobility.
max-idle-time	Creates an idle-duration timer for automatically logging out an Extension Mobility user.
user	Creates an authentication credential to be used by Extension Mobility.

emadmin logout

To manually log out an external application from Extension Mobility, use the **emadmin logout** command in privileged EXEC mode. To return to default, use the **no** form of this command.

emadmin logout *name*
no emadmin logout *name*

Syntax Description

name	Already-configured credential in Extension Mobility user profile.

Command Default

Application remains logged into the Extension Mobility phone until logged out.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command enables an external application, such as a CSTA client application, to log out of an Extension Mobility phone.

Examples

The following example shows how to configure this command to log out an application that logged into an Extension Mobility phone using the “user204” credential:

```
Router# logout user204

Router#
```

Related Commands

Command	Description
user	Creates an authentication credential to be used by Extension Mobility.

emergency response callback

To define a dial peer that is used for 911 callbacks from the PSAP, use the emergency response callback command in voice dial-peer configuration mode. To remove the definition of the dial peer as an incoming link from the PSAP, use the **no** form of this command.

emergency response callback
no emergency response callback

Syntax Description

This command has no arguments or keywords.

Command Default

The dial peer is not defined as an incoming link from the PSAP.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added for Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to define which dial peer is used for 911 callbacks from the PSAP. You can define multiple dial peers with this command.

Examples

The following example shows a dial peer defined as an incoming link from the PSAP. If 408 555-0100 is configured as the ELIN for an ERL, this dial peer recognizes that an incoming call from 408 555-0100 is a 911 callback.

```
dial-peer voice 100 pots
 incoming called-number 4085550100
 port 1/1:D
 direct-inward-dial
 emergency response callback
```

Related Commands

Command	Description
emergency response location	Associates an ERL to either a SIP phone, ephone, or dial peer.
emergency response zone	Defines a dial peer that is used by the system to route emergency calls to the PSAP.
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.

emergency response location

To associate an emergency response location (ERL) for Enhanced 911 Services with a dial peer, ephone, ephone-template, voice register pool, or voice register template, use the **emergency response location** command in dial peer, ephone, ephone-template, voice register pool, or voice register template configuration mode. To remove the association, use the **no** form of this command.

emergency response location *tag*
no emergency response location *tag*

Syntax Description

tag	Unique number that identifies an existing ERL tag defined by the **voice emergency response location** command.

Command Default

No ERL tag is associated with a dial peer, ephone, ephone-template, voice register pool, or voice register template.

Command Modes

Dial-peer configuration (config-dial-peer)
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added to Cisco Unified CME in the ephone-template and voice register template configuration modes.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to assign an ERL to phones individually. Depending on the type of phones (endpoints) that you have, you can assign an ERL to a phone's:

- Dial-peer configuration
- Ephone
- Ephone-template
- Voice register pool
- Voice register template

Note The ephone-template and voice register template modes are not for Cisco Unified SRST or Cisco Unified SIP SRST. Voice register pool mode is not available for Cisco Unified SRST.

These methods of associating a phone with an ERL are alternatives to assigning a group of phones that are on the same subnet as an ERL.

The tag used by this command is an integer from 1 to 2147483647 and refers to an existing ERL tag that is defined by the **voice emergency response location** command. If the tag does not refer to a valid ERL configuration, the phone is not associated to an ERL. For Cisco Unified IP phones, the IP address is used to find the inclusive ERL subnet. For phones on a VoIP trunk or FXS/FXO trunk, the PSAP gets a reorder tone.

Examples

The following example shows how to assign an ERL to a phone's dial peer:

```
dial-peer voice 12 pots
 emergency response location 18
```

The following example shows how to assign an ERL to a phone's ephone:

```
ephone  41
 emergency response location 22
```

The following example shows how to assign an ERL to one or more SCCP phones:

```
ephone-template 6
 emergency response location 8
```

The following example shows how to assign an ERL to a phone's voice register pool:

```
voice register pool 4
 emergency response location 21
```

The following example shows how to assign an ERL to one or more SIP phones:

```
voice register template 4
 emergency response location 8
```

Related Commands

Command	Description
emergency response callback	Defines a dial peer that is used for 911 callbacks from the PSAP.
emergency response zone	Defines a dial peer that is used by the system to route emergency calls to the PSAP.
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.

emergency response zone

To define a dial peer that is used by the system to route emergency calls to a PSAP, use the emergency response zone command in voice dial-peer configuration mode. To remove the definition of the dial peer as an outgoing link to the PSAP, use the **no** form of this command.

emergency response zone *zone-tag*
no emergency response zone

Syntax Description

zone-tag	Identifier (1-100) for the emergency response zone.

Command Default

The dial peer is not defined as an outgoing link to the PSAP. Therefore, E911 services are not enabled.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	The *zone-tag* argument was added and this command was added for Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to specify that any calls using this dial peer are processed by the E911 software. To enable any E911 processing, the emergency response zone command must be enabled under a dial peer.

If no zone tag is specified, the system looks for a matching ELIN to the E911 caller's phone by searching each *emergency response location* that was configured using the **emergency response location** command.

If a zone tag is specified, the system looks for a matching ELIN using sequential steps according to the contents of the configured zone. For example, if the E911 caller's phone has an explicit ERL assigned, the system first looks for that ERL in the zone. If not found, it then searches each location within the zone according to assigned priority numbers, and so on. If all steps fail to find a matching ELIN, the default ELIN is assigned to the E911 caller's phone. If no default ELIN is configured, the E911 caller's automatic number identification (ANI) number is communicated to the Public Safety Answering Point (PSAP).

This command can be defined in multiple dial peers. The zone tag option allows only ERLs defined in that zone to be routed on this dial peer. Also, this command allows callers dialing the same emergency number to be routed to different voice interfaces based on the zone that includes their ERL.

Examples

The following example shows a dial peer defined as an outgoing link to the PSAP. Emergency response zone 10 is created and only calls from this zone are routed through 1/0/0.

```
dial-peer voice 911 pots
 destination-pattern 9911
```

```
prefix 911
emergency response zone 10
port 1/0/0
```

Related Commands

Command	Description
emergency response callback	Defines a dial peer that is used for 911 callbacks from the PSAP.
emergency response location	Associates an ERL to either a SIP phone, ephone, or dial peer.
voice emergency response location	Creates a tag for identifying an ERL for E911 services.
voice emergency response zone	Creates an emergency response zone within which ERLs can be grouped.

encrypt password

To encrypt the password that is configured on Unified CME, use the **encrypt password** command in **telephony-service** configuration mode. To disable password encryption, use the **no** form of this command.

encrypt pasword
no encrypt password

Syntax Description

This command has no arguments or keywords.

Command Default

This command is enabled by default.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is introduced.

Usage Guidelines

The CLI command **encrypt password** is enabled by default on Unified CME router. However, it is mandatory to configure **key config-key password-encrypt** *[key]* and **password encryption aes** along with **encrypt password** to support encryption on Unified CME router.

Note: If the key used to encrypt the password is replaced with a new key (replace key or re-key), then the password is re-encrypted with the new key.

Related Commands

Command	Description
password (auto-register)	Configuress the mandatory password for automatic registration of SIP phones with Unified CME.

ephone

To enter Ethernet phone (ephone) configuration mode for an IP phone for the purposes of creating and configuring an ephone, use the **ephone** command in global configuration mode. To disable the ephone and remove the IP phone configuration, use the **no** form of this command.

ephone *phone-tag*
no ephone *phone-tag*

Syntax Description

phone-tag	Unique sequence number that identifies an ephone during configuration tasks. The maximum number is platform-dependent; refer to Cisco IOS command-line interface (CLI) help.

Command Default

No Cisco IP phone is configured.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

Use the **ephone** command to enter ephone configuration mode. Use ephone configuration mode to create and configure Cisco Unified IP phones in Cisco Unified CME.

Before this command can be used for the first time, you must set the maximum number of ephones using the **max-ephones** command in telephony-service configuration mode. The maximum number of ephones varies by router platform and software version.

Examples

The following example enters ephone configuration mode for a phone with the identifier 4 and assigns ephone-dn 1 to button 1:

```
Router(config)# ephone 4
Router(config-ephone)# button 1:1
```

Related Commands

Command	Description
button	Assigns a button number to the Cisco IP phone directory number.
ephone-dn	Enters ephone-dn configuration mode.
mac-address	Configures the MAC address of a Cisco IP phone.
max-ephones	Configures the maximum number of Cisco IP phones that can be supported by a router.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.

Command	Description
restart all (telephony-service)	Performs a fast reboot of all phones associated with a Cisco CME router.

ephone-dn

To enter ephone-dn configuration mode to configure a directory number for an IP phone line, intercom line, paging line, voice-mail port, or message-waiting indicator (MWI), use the **ephone-dn command in** global configuration mode. To delete an ephone-dn, use the **no** form of this command.

ephone-dn *dn-tag* [{**dual-line** | **octo-line**}]
no ephone-dn *dn-tag*

Syntax Description

dn-tag	Unique number that identifies an ephone-dn during configuration tasks. Range is 1 to the number set by the **max-dn** command.
dual-line	(Optional) Enables two calls per directory number.
octo-line	(Optional) Enables eight calls per directory number.

Command Default

No ephone-dn is configured.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)ZJ	Cisco CME 3.0	The **dual-line** keyword was added.
12.3(4)T	Cisco CME 3.4	The **dual-line** keyword was integrated into Cisco IOS Release 12.3(4)T.
12.4(15)XZ	Cisco Unified CME 4.3	The **octo-line** keyword was added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to enter ephone-dn configuration mode to create directory numbers. In ephone-dn configuration mode, you assign an extension number using the **number** command, a name to appear in the local directory using the **name** command, and other parameters using various commands.

Before using the **ephone-dn** command, you must set the maximum number of ephone-dns to support in your system by using the **max-dn** command. The maximum number of ephone-dns that you can create depends on the software version, router platform, and amount of memory that you have installed.

A dual-line ephone-dn has one virtual voice port and two channels to handle two independent calls. This capacity allows call waiting, call transfer, and conference functions within a single directory number. Dual-line mode is supported on all phone types, but is not appropriate for voice-mail numbers, intercoms, or ephone-dns used for message-waiting indicators, paging, loopback, or hunt groups. Overlays of single-line hunt groups onto dual-line buttons are supported.

An octo-line directory number supports up to eight active calls, both incoming and outgoing, on a single phone button. Unlike a dual-line directory number, which is shared exclusively among phones, an octo-line directory number can split its channels among the phones sharing the directory number. All phones sharing the octo-line directory number are allowed to initiate or receive calls on the idle channels of the directory number.

Ephone-dns are created in single-line mode if the **dual-line** or **octo-line** keyword is not used. To change an ephone-dn from one mode to another, for example from dual-line mode to single-line mode, you must delete the ephone-dn and then re-create it.

Examples

The following example shows how to create directory number 1 with extension 5576:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5576
```

The following example shows an ephone-dn with the number 1001 in dual-line mode. The **no huntstop** command allows calls to continue to hunt to other ephone-dns if this one is busy or does not answer. The **huntstop channel** command disables call hunting to the second channel of this ephone-dn if the first channel is busy or does not answer.

```
Router(config)# ephone-dn 10 dual-line
Router(config-ephone-dn)# number 1001
Router(config-ephone-dn)# no huntstop
Router(config-ephone-dn)# huntstop channel
```

The following example shows an ephone-dn with the number 2001 in octo-line mode. The **huntstop channel** command enables call hunting to up to six channels of this ephone-dn. The remaining two channels are available for outgoing calls or features such as call transfer, call waiting, and conferencing.

```
Router(config)# ephone-dn 20 octo-line
Router(config-ephone-dn)# number 2001
Router(config-ephone-dn)# huntstop channel 6
```

Related Commands

Command	Description
huntstop (ephone-dn and ephone-dn-template)	Disables call hunting for directory numbers or channels.
max-dn	Sets the maximum number of ephone-dns that can be configured.
name	Associates a name with an extension (ephone-dn).
number	Associates a telephone or extension number with a directory number (ephone-dn).

ephone-dn-template

To enter ephone-dn-template configuration mode and create an ephone-dn template containing a standard set of ephone-dn features, use the **ephone-dn-template** command in global configuration mode. To delete an ephone-dn template, use the **no** form of this command.

ephone-dn-template *template-tag*
no ephone-dn-template *template-tag*

Syntax Description

template-tag	Identifier for this ephone-dn template. Range is from 1 to 15.

Command Default

No ephone-dn template is created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to create an ephone-dn template. An ephone-dn template contains a set of ephone-dn attributes that you can easily apply to one or more ephone-dns.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Type **?** in ephone-dn-template configuration mode to see the commands that are available in this mode. The following example shows CLI help for ephone-dn-template configuration mode:

```
Router(config-ephone-dn-template)# ?
Ephone Dn template configuration commands:
  call-forward          Define E.164 telephone number for call forwarding
  call-waiting          Config call-waiting option
  caller-id             Configure port caller id parameters
  corlist               Class of Restriction on dial-peer for this dn
  default               Set a command to its defaults
  description           dn desc, for DN Qualified Display Name
  exit                  Exit from ephone-dn-template configuration mode
  hold-alert            Set Call On-Hold timeout alert parameters
  huntstop              Stop hunting on Dial-Peers
  mwi                   set message waiting indicator options (mwi)
  no                    Negate a command or set its defaults
  pickup-group          set the call pickup group number for the DN
  translate             Translation rule
  translation-profile   Translation profile
```

After creating an ephone-dn template, apply the template to one or more ephone-dns using the **ephone-dn-template** command in ephone-dn configuration mode. Even though you can define up to 15 different ephone templates, you cannot apply more than one template to a particular ephone-dn.

If you try to apply a second ephone-dn template to an ephone-dn that already has a template applied to it, the second template will overwrite the first ephone-dn template configuration after you use the **restart** command to reboot the phone.

To view your ephone-dn-template configurations, use the **show telephony-service ephone-dn-template** command. To see which ephone-dns have templates applied to them, use the **show running-config** command.

Examples

The following example creates ephone-dn template 3, which sets call forwarding on busy and no answer to forward calls to extension 4000 and sets the pickup group to 4. Ephone-dn template 3 is then applied to ephone-dn 23 and ephone-dn 33, which appear on ephones 13 and 14, respectively.

```
ephone-dn-template 3
 call-forwarding busy 4000
 call-forwarding noan 4000 timeout 30
 pickup group 4
ephone-dn 23
 number 2323
 ephone-dn-template 3
ephone-dn 33
 number 3333
 ephone-dn-template 3
ephone 13
 button 1:23
ephone 14
 button 1:33
```

Related Commands

Command	Description
ephone-dn-template (ephone-dn)	Applies an ephone-dn template to an ephone-dn.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco Unified CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service ephone-dn-template	Displays ephone-dn-template configurations.

ephone-dn-template (ephone-dn)

To apply an ephone-dn template to an ephone-dn, use the **ephone-dn-template** command in ephone-dn configuration mode. To remove the ephone-dn template, use the **no** form of this command.

ephone-dn-template *template-tag*
no ephone-dn-template *template-tag*

Syntax Description

template-tag	The template tag for a template created with the **ephone-dn-template** command in global configuration mode. Range is from 1 to 15.

Command Default

No ephone-dn template is applied to the ephone-dn.

Command Modes

Ephone-dn configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **ephone-dn-template** command in ephone-dn configuration mode to apply an ephone-dn template to an ephone. You cannot apply more than one ephone-dn template to an ephone-dn.

If you try to apply a second ephone-dn template to an ephone-dn that already has an ephone-dn template applied to it, the second template will overwrite the first ephone-dn template configuration.

To view your ephone-dn-template configurations, use the **show telephony-service ephone-dn-template** command.

Examples

The following example shows how to create ephone-dn template 3, which sets call forwarding on busy and no answer to forward calls to extension 4000 and sets the pickup group to 4, and apply the template to ephone-dn 23 and ephone-dn 33, which appear on ephones 13 and 14, respectively.

```
ephone-dn-template 3
 call-forwarding busy 4000
 call-forwarding noan 4000 timeout 30
 pickup group 4
ephone-dn 23
 number 2323
 ephone-dn-template 3
ephone-dn 33
 number 3333
 ephone-dn-template 3
ephone 13
 button 1:23
ephone 14
 button 1:33
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Creates an ephone-dn template and enters ephone-dn-template configuration mode.
show telephony-service ephone-dn-template	Displays ephone-dn template configurations.

ephone-hunt

To enter ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco Unified CME system, use the **ephone-hunt** command in global configuration mode. To delete a hunt group, use the **no** form of this command.

ephone-hunt *hunt-tag* {**longest-idle** | **peer** | **sequential**}
no ephone-hunt *hunt-tag*

Syntax Description

hunt-tag	Unique sequence number that identifies the ephone hunt group during configuration tasks. Range is from 1 to 100.
longest-idle	Hunt group in which calls go to the ephone-dn that has been idle the longest.
peer	Hunt group in which the first extension to ring is the number to the right (in the list) of the extension that was the last one to ring when the hunt group was last called. Ringing proceeds in a circular manner, left to right, for the number of hops specified when the ephone hunt group is defined.
sequential	Hunt group in which extensions ring in the order in which they are listed, left to right, when the hunt group is defined.

Command Default

No hunt group is defined.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)T	Cisco CME 3.2	The **longest-idle** keyword was added.
12.4(4)XC	Cisco Unified CME 4.0	The maximum number of hunt groups was increased from 10 to 100.
12.4(9)T	Cisco Unified CME 4.0	This command with the maximum number of hunt groups increased to 100 was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **ephone-hunt** command to enter ephone-hunt configuration mode. Use ephone-hunt configuration mode to create ephone hunt groups in a Cisco Unified CME system.

A hunt group is a list of phone numbers that are assigned to take turns receiving incoming calls for one number, a pilot number that is defined with the **pilot** command. The list of numbers in the hunt group is defined using the **list** command. If a number in the list is busy or does not answer, the call is redirected to the next number in the list. The last number tried is the final number, which is defined using the **final** command.

The order in which the numbers are chosen can be longest-idle, peer, or sequential.

- If the order is longest-idle, each hop is directed to the ephone-dn that has been idle the longest. Idle time is determined from the last time that a phone registered, reregistered, or went on-hook.
- If the order is peer, the first number to which calls are directed is the number to the right of the number in the list that was the last number to ring on the previous occasion that the hunt group was called. If that number is busy or does not answer, the call is directed to the next number in the list and, in the process, circles back to the beginning of the list. In peer hunt groups, the **hops** command specifies how many times a call can hop from number to number before going to the final number, after which the call is no longer forwarded.
- If the order is sequential, the first number to which calls are directed is always the first number in the list. If that number is busy or does not answer, the call is redirected to the next available number in the list, from left to right.

Note If the number of times that a call is redirected to a new number exceeds five, the **max-redirect** command must be used to increase the allowable number of redirects in the Cisco Unified CME system.

To configure a new hunt group, you must specify the **longest-idle**, **peer**, or **sequential** keyword. To change an existing ephone hunt group configuration, the keyword is not required. To change the type of hunt group from peer to sequential or sequential to peer, you must remove the existing hunt group first using the **no** form of the command and then recreate it.

Examples

The following example defines longest-idle ephone hunt group 1 with a pilot number 7501, a final number 8000, and 11 numbers in the list. After a call is redirected six times (makes six hops), it is redirected to the final number 8000.

```
ephone-hunt 1 longest-idle
 pilot 7501
 members logout
 list 7001, 7002, 7023, 7028, 7045, 7062, 7067, 7072, 7079, 7085, 7099
 final 8000
 preference 1
 hops 6
 timeout 20
 no-reg
```

The following example defines peer hunt group number 2. Callers dial the pilot number 5610 to reach the hunt group. The first extension to ring the first time that this hunt group is called is 5601. If 5601 does not answer, the hunt proceeds from left to right, beginning with the extension directly to the right of 5601, for four hops. If none of those extensions answers before the hops limit is reached, the call is forwarded to extension 6000, which is the number for the voice-mail service.

If extension 5601 answers the first call, then the second time someone calls the hunt group, the first extension to ring is 5602. If this call hops until extension 5617 answers it, then the third time someone calls the hunt group, the first extension to ring is 5633. If extension 5633 does not answer, the call is redirected to extension 5601, and so forth.

```
Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 5610
Router(config-ephone-hunt)# members logout
Router(config-ephone-hunt)# list 5601, 5602, 5617, 5633

Router(config-ephone-hunt)# final 6000
```

```
Router(config-ephone-hunt)# hops 4
Router(config-ephone-hunt)# preference 1

Router(config-ephone-hunt)# timeout 30

Router(config-ephone-hunt)# exit
```

The following example defines sequential hunt group number 1. When callers dial extension 5601, the first phone to ring is 5001, then 5002, 5017, and 5028. If none of those extensions answers, the call is forwarded to extension 6000, which is the number for the voice-mail service.

```
Router(config)# ephone-hunt 1 sequential
Router(config-ephone-hunt)# pilot 5601
Router(config-ephone-hunt)# members logout
Router(config-ephone-hunt)# list 5001, 5002, 5017, 5028

Router(config-ephone-hunt)# final 6000
Router(config-ephone-hunt)# preference 1

Router(config-ephone-hunt)# timeout 30

Router(config-ephone-hunt)# exit
```

Related Commands

Command	Description
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco Unified CME system.
members logout	Sets all static members initial state to logout. The **members logout** command is rejected if configured after the **list** command.
no-reg (ephone-hunt)	Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

ephone-hunt login

To authorize an ephone-dn to dynamically join and leave an ephone hunt group, use the **ephone-hunt login** command in ephone-dn configuration mode. To disable this capability, use the **no** form of this command.

ephone-hunt login
no ephone-hunt login

Syntax Description

This command has no arguments or keywords.

Command Default

An ephone-dn is not allowed to dynamically join and leave ephone hunt groups.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **show ephone-hunt** command to display current hunt group members, including those who joined the group dynamically.

Examples

The following example creates five ephone-dns and a hunt group that includes the first two ephone-dns and two wildcard slots. The last three ephone-dns are enabled for group hunt dynamic membership. Each of them can join and leave the hunt group whenever one of the slots is available.

```
ephone-dn 22
 number 4566
ephone-dn 23
 number 4567
ephone-dn 24
 number 4568
 ephone-hunt login
ephone-dn 25
 number 4569
 ephone-hunt login
ephone-dn 26
 number 4570
 ephone-hunt login
ephone-hunt 1 peer
 list 4566,4567,*,*
 final 7777
```

Related Commands

Command	Description
show ephone-hunt	Displays ephone-hunt group configuration, current status, and statistics.

ephone-hunt statistics write-all

Effective with Cisco Unified CME 9.0, the **ephone-hunt statistics write-all** command is replaced by the **hunt group statistics write-all** command in privileged EXEC mode. For more information, see the **hunt group statistics write-all** command.

To write ephone-hunt statistics information to a file, use the **ephone-hunt statistics write-all** command in privileged EXEC mode.

ephone-hunt statistics write-all *location*

Syntax Description

location	The URL or filename to which the statistics should be written.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
15.2(2)T	Cisco Unified CME 9.0	This command was replaced. See the **hunt group statistics write-all** command.

Usage Guidelines

Use this command to write out, in hourly increments, all the ephone hunt group statistics for the past seven days. This command is intended be used when normal hunt group statistics collection is interrupted, perhaps due to TFTP server failure. The commands that are normally used to provide hunt-group statistics are **hunt-group report delay hours**, **hunt-group report every hours**, **hunt-group report url**, and **statistics collect**. These commands allow you to specify shorter, more precise reporting periods and file-naming conventions.

Note Each year on the day that daylight saving time adjusts the time back by one hour at 2 a.m., the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

Examples

The following example writes the ephone hunt group statistics to a file in flash called "huntstats." See the **hunt-group report url** command for explanations of the output fields.

```
Router# ephone-hunt statistics write-all flash:huntstats

Writing out all ephone hunt statistics to tftp now.
11:13:58 UTC Fri Apr 29 2005,
,
01, Fri 11:00 - 12:00, HuntGp, 01, 01, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Fri 12:00 - 13:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Fri 13:00 - 14:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
```

```
01, Fri 14:00 - 15:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Fri 15:00 - 16:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
.
.
.
```

Related Commands

Command	Description
hunt-group report delay hours	Delays hunt-group statistics collection for a specified number of hours.
hunt-group report every hours	Sets the hourly interval at which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco Unified CME B-ACD call statistics are to be sent using TFTP.
show ephone-hunt	Displays ephone hunt group information.
show ephone-hunt statistics	Displays ephone hunt group statistics.
statistics collection	Enables the collection of call statistics for an ephone hunt group.

ephone-template

To create an ephone template to configure a set of phone features and to enter ephone-template configuration mode, use the **ephone-template** command in global configuration mode. To delete an ephone template, use the **no** form of this command.

ephone-template *template-tag*
no ephone-template *template-tag*

Syntax Description

template-tag	Identifier for this ephone template. Range is from 1 to 20.

Command Default

No ephone template is created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The maximum number of templates that can be created was increased from 5 to 20.
12.4(9)T	Cisco Unified CME 4.0	The modification to increase the maximum number of templates that can be created from 5 to 20 was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to create an ephone template containing a set of ephone commands. The template can then be easily applied to one or more ephones.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Type **?** in ephone-template configuration mode to see the commands that are available in this mode and that can be included in an ephone-template. The following example shows CLI help for ephone-template configuration mode at the time that this document was written:

```
Router(config-ephone-template)#?
Ephone template configuration commands:
  after-hour        ephone exempt from after-hour blocking
  codec             Set preferred codec for calls with other phones on this
                    router
  default           Set a command to its defaults
  exit              Exit from ephone-template configuration mode
  fastdial          Define ip-phone fastdial number
  features          define features blocked
  keep-conference   Do not disconnect conference when conference initiator
                    hangs-up.Connect remaining parties together directly using
                    call transfer.
  keepalive         Define keepalive timeout period to unregister IP phone
  keyphone          Identify an IP phone as keyphone
  mtp               Always send media packets to this router
  network-locale    Select the network locale for this template.
  night-service     Define night-service bell
```

```
  no                Negate a command or set its defaults
  paging-dn         set audio paging dn group for phone
  service           Service configuration in ephone template
  softkeys          define softkeys per state
  speed-dial        Define ip-phone speed-dial number
  transfer          transfer related configuration
  transfer-park     customized transfer to park configuration
  transfer-pattern  customized transfer-pattern configuration
  type              Define ip-phone type
  user-locale       Select the user locale for this template.
```

After creating an ephone template, apply the template to one or more ephones using the **ephone-template** command in ephone configuration mode. Even though you can define up to 20 different ephone templates, you cannot apply more than one template to a particular ephone.

After applying a template to an ephone or removing a template from an ephone, use the following commands:

- **restart**—Performs a fast reboot of the phone.
- **create cnf-files**—Rebuilds configuration files.

If you try to apply a second ephone template to an ephone that already has an ephone template applied to it, the second template will overwrite the first ephone template configuration after you use the **restart** command to reboot the phone.

To view your ephone-template configurations, use the **show telephony-service ephone-template** command. To see which ephones have templates applied to them, use the **show running-config** command.

Examples

The following example creates two ephone templates. The **softkeys** commands in ephone-template configuration mode define what soft keys are displayed and their order. Template 1 is applied to ephone 32, which has the extension 2555, and template 2 is applied to ephone 38, which has the extension 2666.

```
ephone-template 1
 softkeys idle Dnd Redial Newcall Pickup Login
 softkeys seized Redial Cfwdall Gpickup Pickup
 softkeys alerting Callback Endcall
 softkeys connected Confrn Hold Endcall
ephone-template 2
 softkeys idle Redial Pickup
 softkeys seized Redial Pickup
 softkeys connected Hold Endcall
ephone-dn 25
 number 2555
ephone-dn 26
 number 2666
ephone 32
 button 1:25
 ephone-template 1
ephone 38
 button 1:26
 ephone-template 2
```

The following example creates an ephone template to block the use of Park and Trnsfer soft keys. It is applied to extension 2333.

```
ephone-template 15
 features blocked Park Trnsfer
ephone-dn 2
 number 2333
```

```
ephone 3
 button 1:2
 ephone-template 15
```

Related Commands

Command	Description
create cnf-files	Builds phone configuration files.
ephone-template (ephone)	Applies an ephone template to an ephone.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco Unified CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service ephone-template	Displays ephone-template configurations.

ephone-template (ephone)

To apply an ephone template to a particular SCCP phone in Cisco Unified CME, use the **ephone-template** command in ephone configuration mode. To remove the ephone template, use the **no** form of this command.

ephone-template *template-tag*
no ephone-template *template-tag*

Syntax Description

template-tag	Unique identifier for a template created by using the **ephone-template** command in global configuration mode. Range is 1 to 20.

Command Default

No ephone template is applied to a phone.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified 4.0	The maximum number of ephone templates that can be created was increased from 5 to 20.
12.4(9)T	Cisco Unified 4.0	This command with an increased range for the *template-tag* argument was integrated into Cisco IOS Release 12.4(9)T.
12.4(15)XZ	Cisco Unified CME 4.3	This command was modified to specify that before an ephone template can be applied to a particular phone, the Mac address for that phone must be present in its configuration file.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command in ephone configuration mode applies an ephone template to a particular phone.

In Cisco Unified CME 4.3 and later versions, an ephone template cannot be applied to a particular phone unless its configuration file includes its MAC address. If you attempt to apply a template to a phone for which the MAC address in not configured, a message appears. To configure the MAC address for a Cisco Unified IP phone, use the **mac-address** command in ephone configuration mode.

After applying an ephone template, use the **restart** command to perform a fast reboot of the phone.

You cannot apply more than one ephone template at a time to any phone. If you attempt to apply a second ephone template to phone to which an ephone template is already applied, the second template will overwrite the first ephone template configuration after you reboot the phone.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value set in ephone configuration mode has priority over the value set in ephone-template configuration mode.

To view your ephone-template configurations, use the **show telephony-service ephone-template** command.

Examples

The following example defines ephone templates 1 and 2 and applies ephone template 1 to ephones 1 through 3 and ephone template 2 to ephone 4.

```
ephone-template 1
 softkeys idle Dnd Redial Newcall Pickup Login
 softkeys seized Redial Cfwdall Gpickup Pickup
 softkeys alerting Callback Endcall
 softkeys connected Confrn Hold Endcall
 softkeys hold Newcall Resume
ephone-template 2
 softkeys idle Redial Pickup
 softkeys seized Redial Pickup
 softkeys alerting Endcall
 softkeys connected Hold Endcall
 softkeys hold Resume
 ephone 1
 ephone-template 1
ephone 2
 ephone-template 1
 ephone 3
 ephone-template 1
ephone 4
 ephone-template 2
ephone 5
 ephone-template 2
The following example creates an ephone template to block the use of Park and Transfer soft
 keys on extension 2333.
ephone-template 15
 features blocked Park Trnsfer
ephone-dn 2
 number 2333
ephone 3
 button 1:2
 ephone-template 15
```

Related Commands

Command	Description
ephone-template	Creates an ephone-template and enters ephone-template configuration mode.
mac-address	Associates the MAC address of a Cisco Unified IP phone with an ephone configuration in Cisco Unified CME.
restart (ephone)	Performs a fast reboot of a single phone in Cisco Unified CME.
restart (telephony-service)	Performs a fast reboot of one or all phones in Cisco Unified CME.
show telephony-service ephone-template	Displays ephone-template configurations.

ephone-type

To add a Cisco Unified IP phone type by defining an ephone-type template, use the **ephone-type** command in global configuration mode. To remove an ephone type, use the **no** form of this command.

ephone-type *phone-type* [**addon**]
no ephone-type *phone-type*

Syntax Description

phone-type	Unique label that identifies the type of phone. Value is any alphanumeric string up to 32 characters.
addon	(Optional) Phone type is an add-on module, such as the Cisco Unified IP Phone 7915 Expansion Module.

Command Default

Ephone type is not defined.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unifieid SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unifieid SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command adds a user-defined template for a phone type to a Cisco Unified CME system. This configuration template defines a set of attributes that describe the features of the new phone type. Use this command to add phone types that are not already defined in the system.

If you use a phone-type argument that matches a system-defined phone type, a message displays notifying you that the ephone-type is built-in and cannot be changed. For a list of system-defined phone types, see the **type** command.

Use the **create cnf-files** command for the new phone type to take effect.

Examples

The following example shows the Nokia E61 added with an ephone-type template, which is then assigned to ephone 2:

```
ephone-type  E61
 device-id 376
 device-name E61 Mobile Phone
 num-buttons 1
 max-presentation 1
 no utf8
 no phoneload
!
!
telephony-service
 load E61 SCCP61.8-2-2SR2S
 max-ephones 100
```

```
 max-dn 240
 ip source-address 15.7.0.1 port 2000
 cnf-file location flash:
 cnf-file perphone
 voicemail 8900
 max-conferences 8 gain -6
 transfer-system full-consult
 create cnf-files version-stamp 7960 Sep 25 2007 21:25:47
!
!
ephone  2
 mac-address 001C.821C.ED23
 type E61
 button  1:2
```

Related Commands

Command	Description
create cnf-files	Builds the eXtensible Markup Language (XML) configuration files that are required for IP phones.
device-id	Specifies the device ID for a phone type in an ephone-type template.
device-name	Assigns a name to a phone type in an ephone-type template.
load	Associates a type of Cisco Unified IP phone with a phone firmware file.
type	Assigns a phone type to an SCCP phone.

exclude

To exclude the availability of local services on a phone's user interface such as, Extension Mobility (EM), My Phone Apps, and Local Directory from the phone's configuration, use the exclude command in ephone or ephone-template mode.

exclude [{**em** | **myphoneapp** | **directory** | **call-history**}]

Syntax Description

em	Extension mobility (EM) service.
myphoneapp	My Phone Apps service.
directory	Local directory service
call-history	Call history in the missed, received, or placed calls directory

Command Default

Local services are enabled.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.
15.1(4)M	Cisco Unified CME 8.6	This command was modified. Call-history option was added.

Usage Guidelines

Use this command to exclude the availability of local services such as, EM, my phone apps, and local directory services from the phone configuration.

Examples

The following example shows directory and my phone apps excluded from ephone-template 8:

```
Router# conf t
Router#config)#ephone-template 8
Router(config-ephone-template)#exclude ?
  directory   local directory service
  em          extension mobility service
  myphoneapp  my phone apps service
  <cr>
Router(config-ephone-template)#exclude directory
Router(config-ephone-template)#exclude myphoneapp!
```

The following example shows call-history as excluded from ephone 10:

```
!
telephony-service
 max-ephones 40
 max-dn 100
 max-conferences 8 gain -6
 transfer-system full-consult
!
!
```

```
ephone-template  5
 exclude call-history
!
!
ephone  10
 exclude call-history
 device-security-mode none
!
```

Related Commands

Command	Description
ephone-template (ephone)	Applies template to an ephone.
show telephony-service ephone-template	Displays ephone-template configurations.

exclude (voice register)

To exclude from the Cisco Unified SIP IP phone's user interface the availability of local services such as Extension Mobility (EM), My Phone Apps, and Local Directory, use the **exclude** command in voice register pool or voice register template configuration mode.

exclude [{**em** | **myphoneapps** | **directory**}]

Syntax Description

em	Extension Mobility service is excluded.
myphoneapps	My Phone Apps service is excluded.
directory	Local Directory service is excluded.

Command Default

Local services are enabled.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Examples

The following example shows the Local Directory and My Phone Apps services excluded from voice register pool 33:

```
Router(config)# voice register pool 33
Router(config-register-pool)# exclude directory
Router(config-register-pool)# exclude myphoneapps
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.

expiry

To set the time after which emergency callback history expires, use the **expiry** command in voice emergency response settings configuration mode. To remove the number, use the **no** form of this command.

expiry *time*
no expiry

Syntax Description

time	Identifier (2-2880) in minutes for the maximum time the 911 caller history is available for callback.

Command Default

The default expiry time is 180 minutes.

Command Modes

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to specify the amount of time (in minutes) to keep emergency caller history for each ELIN. The time can be an integer in the range of 2 to 2880 minutes. The default value is 180 minutes.

Examples

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller's IP phone address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500.

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
```

Related Commands

Command	Description
callback	Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.
elin	E.164 number used as the default ELIN if no matching ERL to the 911 caller's IP phone address is found.
logging	Syslog informational message printed to the console every time an emergency call is made.

Command	Description
voice emergency response settings	Creates a tag for identifying settings for E911 behavior.

extension-assigner tag-type

To enable provision tags for identifying ephone configurations when using the extension assigner application, use the **extension-assigner tag-type** command in telephony-service configuration mode. To return to the default setting of using the ephone tag, use the **no** form of this command.

extension-assigner tag-type {**ephone-tag** | **provision-tag**}
no extension-assigner tag-type

Syntax Description

ephone-tag	Ephone tags must be used to identify ephone configurations.
provision-tag	Provision tags must be used to identify ephone configurations.

Command Default

Ephone tags are used to identify ephone configurations for the extension assigner application.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command enables you to use provision tags for identifying ephone configurations to be assigned by the extension application application.

A provision tag is an unique number other than an ephone tag, such as a jack number or an extension number, for identifying the ephone configuration to be assigned to a particular IP phone by the extension assigner application.

Use this command to specify which type of tag, ephone tag or provision tag, is to be used to identify ephone configurations for the extension assigner application. The default configuration is ephone tag.

If you use this command with the **provision-tag** keyword**,** use the **provision-tag** command to create provision tags.

Examples

The following example shows that this command is configured to enable provision tags to be used for identifying the ephone configurations to be assigned by the extension assigner application. Note that provision tag 1001 is configured for ephone 1. During phone installation, the installation technician can press 1001 on the telephone keypad to assign the ephone 1 configuration, with extension number 1001 on button 1, to the IP phone being installed.

```
Telephony-service
 extension-assigner tag-type provision-tag
 auto assign 101-102
 auto-reg-ephone
Ephone-dn 101
 number 1001
```

```
Ephone-dn 102
 number 1002
Ephone 1
 provision-tag 1001
 mac-address 02EA.EAEA.0001
 button 1:101
Ephone 2
 provision-tag 1002
 mac-address 02EA.EAEA.0002
 button 1:102
```

Related Commands

Command	Description
provision-tag	Creates a provision tag for identifying an ephone configuration.

extension-range

To define a range of extension numbers for a specific MOH group in Cisco Unified CME or Cisco Unified SRST, use the **extension-range** command in voice-moh-group configuration mode. To define a range of extension numbers for a specific directory number in Cisco Unified CME, use the **extension-range** command in ephone-dn configuration mode. To disable the extension-range command, use the **no** form of this command.

extension-range *starting-extension* *to* *ending-extension*
no **extension-range** *starting-extension* **to** *ending-extension*

Syntax Description

starting-extension	Hexadecimal digits (0-9 or A-F) that define the starting extension number in an extension range. Maximum length: 32 digits.
ending-extension	Hexadecimal digits (0-9 or A-F) that define the last extension number in an extension range. Value of the ending extension must be larger than value of the starting extension. Maximum length: 32 digits.

Command Default

No extension- range is configured.

Command Modes

Voice MOH group configuration (config-voice-moh-group)
Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command configured in voice moh-group configuration mode identifies MOH clients calling extension numbers specified under the extension range configured for a MOH group in Cisco Unified CME or Cisco Unified SRST. This command in ephone-dn configuration mode identifies MOH clients calling extension numbers specified under the extension range configured for a directory number in Cisco Unified CME

You can define multiple extension-ranges in the same MOH group or directory number.

The extension can be expressed in hexadecimal digits ranging from 0-9 or A-F and should not exceed the limit of 32 digits.

The starting-extension and ending-extension numbers must contain the same number of digits.

The ending extension number must be of a greater value than the starting extension number.

Extension-range for a MOH group must not overlap with any other extension-range configured in any other MOH group.

Note If an extension range is defined in a MOH group and it is also defined under ephone-dn, the extension range defined under ephone-dn takes precedence.

Examples

The following example shows two extension ranges configured under voice moh-group 1:

```
Router(config)# voice moh-group 1
Router(config-voice-moh-group)# moh flash:/minuet.wav
Router(config-voice-moh-group)# description Marketing
Router(cconfig-voice-moh-group)# extension range 1000 to 1999
Router(config-voice-moh-group)# extension range 3000 to 3999
```

Related Commands

Command	Description
moh	Enables music on hold from an audio file.
voice-moh-group	Enters voice moh-group configuration mode.

external-ring (voice register global)

To specify the type of ring sound used on Cisco Session Initiation Protocol (SIP) or Cisco SCCP IP phones for external calls, use the **external-ring** command in voice register global configuration mode. To return to the default ring sound, use the **no** form of this command.

external-ring {**bellcore-dr1** | **bellcore-dr2** | **bellcore-dr3** | **bellcore-dr4** | **bellcore-dr5**}
no external-ring

Syntax Description

bellcore-dr1 bellcore-dr2 bellcore-dr3 bellcore-dr4 bellcore-dr5	Standard distinctive ringing patterns as defined in the standard GR-506-CORE, *LSSGR: Signaling for Analog Interfaces* .

Command Default

The default ring sound is an internal ring pattern.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

When set, this command defines varying ring tones so that you can discriminate between internal and external calls from Cisco SIP or Cisco SCCP IP phones.

Examples

The following example shows how to specify that Bellcore DR1 be used for external ringing on Cisco SIP IP phones:

```
Router(config)# voice register global
Router(config-register-global)# external-ring bellcore-dr1
```

Cisco Unified CME Commands: F

fac

To enable all standard feature access codes (FACs) or to create and enable individual custom FACs, use the **fac** command in telephony-service configuration mode. To disable FACs, use the **no** form of this command.

fac {**standard** | **custom** { **alias** *alias-tag feature* } }

fac refer
no fac { {**standard** | **custom** {**alias** *alias-tag feature* }

Syntax Description

standard	All predefined standard FACs are enabled.
custom	User-defined FAC for selecting a particular feature or function from the predefined set of features is enabled.
alias	Alternative FAC for dialing an existing FAC or existing FAC plus extra digits without removing the existing FAC is enabled.
alias-tag	Unique number that identifies this alias during configuration tasks. Range: 0 to 9.
custom-fac	User-defined code to dial using the keypad on an IP or analog phone. Code can be up to 256 characters and can contain numbers 0 to 9 and * and #. **Note** ## is not supported for FACs on SIP phones.
to	Maps custom FAC being configured to specified target.
existing-fac	Already configured custom FAC that is automatically dialed when the phone user dials the custom FAC being configured.
extra-digits	(Optional) Additional digits that are automatically dialed when the phone user dials the custom FAC being configured. Valid entries are: • **target extension** —Telephone or extension number in Cisco Unified CME to which the incoming calls are forwarded. Used with the Call Forward feature. • **group number** —Pickup group number, for a group other than the local group number. Used with the Pickup Group feature. • **pickup extension** —Telephone or extension number in Cisco Unified CME to be picked up when ringing. To be used with the Pickup Direct feature. • **park-slot number** —Number on which calls are to be temporarily parked. Use with the Call Park feature. Target park slot must be already configured in Cisco Unified CME. • **pilot number** —Telephone or extension number configured as a the pilot number for an ephone hunt group to be joined. Hunt group to be joined must allow dynamic membership.

feature	Predefined alphabetic string that identifies a particular feature or function. Valid entries are: • **callfwd all** —Directs system to forward all incoming calls for this telephone or extension number. • **callfwd cancel** —Directs system to cancel the call-forward-all selection. • **ccw** —Disables the Call Waiting feature. • **dnd** —Enables Do Not Disturb (DND) feature on SCCP phones. Not supported for SIP phones. • **dpark-retrieval** —Enables Directed Call Park Retrieval feature. Applies to both SIP and SCCP phones. • **ephone-hunt cancel** —Leaves an ephone hunt group that is configured to allow dynamic membership. • **ephone-hunt hlog** —Activates or deactivates hunt group logout functionality, changing the status of the an ephone-dn for a hunt group agent from ready to not-ready or from not-ready to ready. • **ephone-hunt hlog-phone** —Activates or deactivates phone-level hunt group logout functionality, changing the status of all the extensions on a hunt group member phone from ready to not-ready or from not-ready to ready. • **ephone-hunt join** —Joins an ephone hunt group that is configured to allow dynamic membership. If multiple hunt groups have been created that allow dynamic membership, the hunt group to be joined is identified by its pilot number. • **park** —Enables Call Park feature. • **pickup direct** —Picks up a ringing call at any extension. Applies to both SIP and SCCP phones. • **pickup group** —Picks up a ringing call in a different pickup group than yours. Applies to both SIP and SCCP phones. • **pickup local** —Picks up a ringing call in your pickup group. Applies to both SIP and SCCP phones. • **redial** —Redials the last number called. • **trnsfvm** —Activates the Transfer to Voice-Mail feature. • **voicemail** —Dials the voice-mail number.

Command Default

FACs are disabled on IP phones.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(15)XZ	Cisco Unified CME 4.3	Standard FAC and **trnsfvm** keyword for a custom FAC were added for transfer to voice mail.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	The **dpark-retrieval** keyword was added and support for SIP phones was added for the **park direct**, **park group**, and **park local** keywords.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. The **ccw** keyword was added for a custom FAC.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use this command to enable all predefined standard FACs or to create one or more custom FACs.

FACs enable phone users to use the keypad on an analog or IP phone registered in Cisco Unified CME to select or activate/deactivate a particular feature or function from a predefined set of features. For example, a phone user might press **1, then press 2345 to forward all incoming calls to extension 2345.

Standard FACs and custom FACs are mutually exclusive. You can enable all standard FACs or create and enable one or more custom FACs.

Most FACs are valid only immediately after a phone user goes off-hook. The only exception is the call-park FAC. The call-park FAC actually invokes a call transfer to a park slot. To use the call-park FAC, a phone user must have an active call and must press the Transfer soft key (IP phone) or hookflash (analog phone) before dialing the call-park FAC. Dialing the FAC for the Call Park feature does not use the Park soft key function.

Use the **fac standard** command to enable all predefined standard FACs for all SCCP phones registered in Cisco Unified CME.

Use the **fac custom** command to create an individual custom FAC for selecting a particular feature or function from the predefined feature set.

Use the **fac custom** command with the **alias** keyword to create an alternative (custom) FAC for dialing an existing FAC, or existing FAC plus extra digits without removing the existing FAC. For example, an alias can be created to allow the phone user to press **1 to forward all incoming calls to a particular extension *without* requiring the phone user to dial the target extension number.

To disable *all* custom FACs, use the **fac standard** command, which enables all standard FACs. To disable all standard FACs or to disable an individual custom FAC, use the **no** form of the **fac** command.

Use the **show telephony-service fac** command to display a list of FACs that are configured on the Cisco Unified CME router.

Examples

The following example shows how to enable standard FACs for all phones:

```
Router(config)# telephony-service
Router(config-telephony)# fac standard
fac standard is set!
```

The following example shows the output from the **show telephony-service fac** command when standard FACs are enabled:

```
Router# show telephony-service fac

telephony-service fac standard
 callfwd all **1
 callfwd cancel **2
 pickup local **3
 pickup group **4
 pickup direct **5
 park **6
 dnd **7
 redial **8
 voicemail **9
 ephone-hunt join *3
 ephone-hunt cancel #3
 ephone-hunt hlog *4
 ephone-hunt hlog-phone *5
 trnsfvm *6
 dpark-retrieval **10
 cancel call waiting *1
```

The following example shows how the standard FAC for the Call Forward All feature is changed to a custom FAC (#45). Then an alias is created to map a second custom FAC to #45 plus an extension (1111). The second custom FAC (#44) allows the phone user to press #44 to forward all calls all calls to extension 1111, without requiring the phone user to dial the extra digits that are the extension number.

```
Router(config)# telephony-service
Router(config-telephony)# fac custom callfwd all #45
fac callfwd all code has been configured to #45
Router(config-telephony)# fac custom alias 0 #44 to #451111
fac alias0 code has been configurated to #44!
alias0 map code has been configurated to #451111!
```

The following example shows how to create three aliases for the Group Pickup feature. The FAC for group pickup is **4. The three new custom FACs are #1, #2, and #4 to pickup groups 121, 122, and 124, respectively. This allows a phone user to press #1 to pick up calls in group 121, #2 to pick up calls in group 122, and #4 to pick up calls in group 124.

```
Router(config)# telephony-service
Router(config-telephony)# fac custom pickup group **4
fac pickup group code has been configured to **4
Router(config-telephony)# fac custom alias 1 #1 to **4121
fac alias1 code has been configurated to #1!
alias1 map code has been configurated to **4121!
Router(config-telephony)# fac custom alias 2 #2 to **4122
fac alias2 code has been configurated to #2!
alias2 map code has been configurated to **4122!
Router(config-telephony)# fac custom alias 4 #4 to **4124
fac alias4 code has been configurated to #4!
alias4 map code has been configurated to **4124!
```

The following example shows the output from the **show telephony-service fac** command when custom FACs are configured:

```
Router# show telephony-service fac
```

```
telephony-service fac custom
 callfwd all #45
 alias 0 #44 to #451111
 alias 1 #1 to **4121
 alias 2 #2 to **4122
 alias 4 #4 to **4124
```

Related Commands

Command	Description
show telephony-service fac	Displays list of FACs that are configured on Cisco Unified CME.

fac refer

To send the SIP REFER to a SIP phone. use the **fac refer** command in voice register global configuration mode. To allow Cisco Unified CME to handle the SIP REFER internally, use the **no** form of this command.

fac refer
no fac refer

Syntax Description

lpcor-group	Name of the LPCOR resource group.

Command Default

Fac refer is enabled.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this ommand to control the SIP REFER to be sent to a SIP phone. The fac refer command is enabled in Cisco Unified CME by default to allow Cisco Unified CME to pass the REFER to the SIP phone, thereby enabling the phone to make a new call towards Cisco Unified CME . Cisco Unified CME accepts the new invite message as a new call and requires the call transferree to enter a forced authorization code (FAC) again.

Use the no fac refer command to allow Cisco Unified CME to handle the SIP REFER internally instead of passing the call towards the SIP phone.

Examples

The following example shows no fac refer configured in voice register global:

```
Router#show run
!
voice register global
 no fac refer
```

Related Commands

Command	Description
show voice register global	Displays all global configuration parameters associated with SIP phones.

fail-connect-time

To specify the maximum time to wait for establishing VPN tunnel including establishing of SSL/DTLS and login or connect requests or responses, use the **fail-connect-time** command in vpn-profile configuration mode. To disable the fail-connect-time configuration, use the no form of this command.

fail-connect-time seconds

Syntax Description

seconds	Failure-to-connect time, in seconds. Range: 0 to 600 seconds. Default: 30 seconds.

Command Default

Default fail-connect-time is 30 seconds.

Command Modes

Vpn-profile configuration (conf-vpn-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to specify the fail-to-connect time for a vpn-profile. The fail-to-connect time specifies the maximum time to wait for establishing VPN tunnel including establishing of SSL/DTLS and login/connect request/response. The fail-to-connect time ranges from 0 seconds to 600 seconds. The default fail-to-connect time is 30 seconds.

Examples

The following example shows fail-connect-time set to 50 seconds for vpn-profile 4:

```
Router# show run
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  keepalive 50
  host-id-check disable
 vpn-profile 2
  mtu 1300
  password-persistent enable
  host-id-check enable
 vpn-profile 4
  fail-connect-time 50
 sip
!
```

Related Commands

Command	Description
vpn-profile	Defines a VPN-profile.

fastdial

To create an entry for a personal speed-dial number, use the **fastdial command in** ephone or ephone-template configuration mode. To delete a personal speed-dial number, use the **no** form of this command.

fastdial *dial-tag number* **name** *name-string*
no fastdial *dial-tag*

Syntax Description

dial-tag	Unique sequence number that ranges from 1 to 100 and is used to identify a particular personal speed-dial number during configuration tasks.
number	Telephone number or extension to be dialed.
name *name-string*	Label to appear in the Personal Speed Dial menu, containing a string of up to 24 alphanumeric characters. Personal speed dial is handled through an XML request, so characters that have special meaning to HTTP, such as ampersand (&), percent sign (%), semicolon (;), angle brackets (< >), and vertical bars (||), are not allowed.

Command Default

No personal speed-dial numbers are present.

Command Modes

Voice register pool configuration (config-register-pool)
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.
15.4(3)M	Cisco Unified CME 10.5	This command was modified to increase the range from 24 to 100.

Usage Guidelines

This command is supported only on certain Cisco Unified IP phones, such as the 7940, 7960, 7960G, 7970G, and 7971G-GE. To determine whether personal speed-dial menu is supported on your IP phone, see the Cisco Unified CME user documentation for your IP phone model.

Phone users access personal speed-dial numbers through the Directories > Local Services > Personal Speed Dial menu. Personal speed-dial numbers appear on this menu in the order in which they are entered during configuration.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example creates a directory of five personal speed-dial numbers for an IP phone:

```
Router(config)# ephone 1
Router(config-ephone)# fastdial 1 5001 name Front Register
Router(config-ephone)# fastdial 2 5002 name Security
Router(config-ephone)# fastdial 3 5003 name Rear Register
Router(config-ephone)# fastdial 4 5004 name Office
Router(config-ephone)# fastdial 5 912135550122 Accounting
```

Related Commands

Command	Description
ephone-template (ephone)	Applies a template to the ephone being configured.
show telephony-service ephone-template	Displays the contents of ephone templates.

feature-button

To enable feature button configuration on a line key, use the feature-button command in ephone, ephone-template, voice user profile, or voice logout profile configuration mode. To disable the feature button configuration on a line key, use the no form of this command.

feature-button *index* *index* <*feature identifier*> **[label** <*label*>**]**
no feature-button *index* *index* <*feature identifier*> **[label** <*label*>**]**

Syntax Description

index	Index number of a specific feature type. One from the total 24 feature IDs.
feature identifier	One of the following feature or stimulus IDs: Redial, Hold, Trnsfer, Cfwdall, Privacy, MeetMe, Confrn, Park, Pickup. Gpickup, Mobility, Dnd, ConfList, RmLstC, CallBack, NewCall, EndCall, HLog, NiteSrv, Acct, Flash, Login, TrnsfVM, LiveRcd.
label	Defines non-default text label for PLK button.

Command Default

No **feature-button** is configured.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Voice user-profile configuration (config-user-profile)
Voice logout-profile configuration (config-logout-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(3)T	Cisco Unified CME 8.5	This command was modified to configure feature button on a phone’s line key. Feature button index number and feature ID keywords were added.
15.2(4)M	Cisco Unified CME 9.1	This command was modified to add **label** <*label*> for the PLK button.

Usage Guidelines

Use this command to configure a DnD feature button as a short cut for the DnD softkey. This command with the **privacy** keyword takes precedence over the **privacy-button** command. If a **feature-button** is configured for DnD, the **privacy-button** command will be ignored and the privacy button must be configured through the feature-button command to take effect.

In Cisco Unified CME 8.5 and later versions, the feature-button command allows you to program a phone’s line key to function as a feature button. You can configure one of the following 24 feature IDs: Redial, Hold, Trnsfer, Cfwdall, Privacy, MeetMe, Confrn, Park, Pickup. Gpickup, Mobility, Dnd, ConfList, RmLstC, CallBack, NewCall, EndCall, HLog, NiteSrv, Acct, Flash, Login, TrnsfVM, LiveRcd

Examples

The following example shows how to configure feature buttons:

```
Router(config)# ephone 1
Router(config-ephone) feature-button 1 privacy
Router(config-ephone) feature-button 2 dnd
Router(config-ephone) feature-button 3 Hlog label Agent Hlogout

The following example shows feature buttons configured in ephone template 9 and ephone
template 10:
Router# show telephony-service ephone-template
ephone-template 9
conference drop-mode never
conference add-mode all
conference admin: No
max-calls-per-button 8
busy-trigger-per-button 0
privacy default
feature-button 1 Endcall
feature-button 3 Mobility
Always send media packets to this router: No
Preferred codec: g711ulaw
keepalive 30 auxiliary 30
User Locale: US
Network Locale: US
lpcor type:
lpcor (incoming):        (outgoing):
ephone-template 10
conference drop-mode never
conference add-mode all
conference admin: No
max-calls-per-button 8
busy-trigger-per-button 0
privacy default
feature-button 1 Park
feature-button 2 MeetMe
feature-button 3 CallBack
button-layout 1 line
button-layout 2-4 speed-dial
button-layout 5-6 blf-speed-dial
MLPP Service Domain Network none (0)
!
```

Related Commands

Command	Description
privacy-button	Enables the privacy feature button on an IP phone.
show telephony-service ephone	Displays the information about ephone configuration in a Cisco CallManager Express (Cisco CME) system.
show telephony-service ephone-dn-template	Displays the information about ephone-template's configurations.

feature-button (voice_register_pool)

To configure feature button configuration on a line key, use the feature-button command in voice register pool or voice register template configuration mode. To disable the feature button configuration on a line key, use the no form of this command.

feature-button [**index number feature identifier feature id**]
no feature button [**index number feature identifier feature id**]

Syntax Description

index	Index number of a specific feature type. One from the total 24 feature IDs.
feature identifier	One of the following feature or stimulus IDs: Redial, Hold, Trnsfer, Cfwdall, Privacy, MeetMe, Confrn, Park, Pickup, Gpickup, Mobility, NewCall, EndCall, Dnd, ConfList, NewCall, HLog, Trnsfer.

Command Default

Feature-button configuration on a line key is disabled.

Command Modes

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to program a phone's line key to function as a feature button. You can configure one of the following 24 features IDs: Redial, Hold, Trnsfer, Cfwdall, Privacy, MeetMe, Confrn, Park, Pickup, Gpickup, Mobility, NewCall, EndCall, Dnd, ConfList, NewCall, HLog, Trnsfer. The feature ID list for the command is incrementally updated across Unified CME releases.

Examples

The following example shows feature button configured in voice register pool 50:

```
voice register pool  50
 id mac 001E.7AC4.DC73
 feature-button 1 NewCall
 type 7965
 number 1 dn 65
 template 1
 dtmf-relay rtp-nte
 speed-dial 1 2001 label "SD1-2001"
 speed-dial 3 2003 label "SD3-2003"
 blf-speed-dial 1 3001 label "BLFl1-3001"
!
```

Related Commands

Command	Description
show voice register pool	Displays all configuration information associated with a particular voice register pool.

features blocked

To prevent one or more features from being used on a Cisco Unified CME phone, use the **features blocked** command in ephone-template configuration mode. To allow all features to be used, use the **no** form of this command.

features blocked [**CFwdAll**] [**Confrn**] [**GpickUp**] [**Park**] [**PickUp**] [**Trnsfer**]
no features blocked

Syntax Description

CFwdAll	Call forward all calls.
Confrn	Conference.
GpickUp	Group call pickup.
Park	Call park.
PickUp	Directed or local call pickup. This includes pickup last-parked call and pickup from another extension or park slot.
Trnsfer	Call transfer.

Command Default

Features are not blocked.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to specify one or more features to be blocked in an ephone template, then apply the template in ephone configuration mode to one or more ephones to prevent the use of the specified features by those ephones. This feature can be used on IP phones and analog phones. After applying the template, any soft keys associated with the blocked features will still be visible but will not have any effect.

Use the **show telephony-service ephone-template** command to display the contents of ephone templates.

Examples

In the following example, call park and call transfer are blocked on ephone 3.

```
ephone-template 1
 features blocked Park Trnsfer
ephone-dn 2
 number 2333
ephone 3
 button 1:2
 ephone-template 1
```

The following example blocks the use of the conference feature on ephone 3, which is an analog phone, by using a template.

```
ephone-template 1
 features blocked Confrn
ephone-dn 78
 number 2579
ephone  3
 ephone-template 1
 mac-address C910.8E47.1282
 type anl
 button  1:78
```

Related Commands

Command	Description
ephone-template (ephone)	Applies a template to the ephone being configured.
show telephony-service ephone-template	Displays the contents of ephone templates.

feed

To enable an audio stream for multicast from a external live audio feed connected directly to the router by a foreign exchange office (FXO) or an E&M analog voice port, use the **feed** command in ephone-dn configuration mode. To disable the multicast audio stream, use the **no** form of this command.

feed ip *ip-address* **port** *port-number* [**route** *ip-address*] [**out-call** *outcall-number*]
no feed ip

Syntax Description

ip *ip-address*	Indicates that a particular audio stream is to be used as a multicast source and specifies the destination IP address for multicast.
port *port-number*	Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CallManager Express (Cisco CME) router.
route *ip-address*	(Optional) Indicates the specific router interface on which to transmit the IP multicast packets. The default is that the audio stream is automatically output on the interface that corresponds to the address that was configured with the **ip source-address** command.
out-call *outcall-number*	(Optional) Sets up a call to the outcall number in order to connect to a live audio feed. If this keyword is not used, the live feed is assumed to derive from an incoming call to the ephone-dn that is being configured.

Command Default

No multicast audio stream is enabled on an extension.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

When this command is used, a connection for a live feed audio stream is established as an automatically connected voice call. If the **out-call** keyword is used, the Cisco CME system calls out to the specified number for the audio stream. If the **out-call** keyword is not used, it is assumed that the call is incoming to the ephone-dn. This includes VoIP calls if voice activity detection (VAD) is disabled. The typical operation is for the Cisco CME ephone-dn to establish a call to a local router E&M voice port.

Connection via E&M is the recommended mechanism because it requires minimal external components. The E&M port must be placed in 4-wire operation, using E&M immediate signaling and with the auto-cut-through option enabled. You directly connect a line-level audio feed (standard audio jack) to pins 3 and 6 of an E&M RJ-45 connector. The E&M WAN interface card (WIC) has a built-in audio transformer that provides appropriate electrical isolation for the external audio source. (The audio connection on the E&M port does not require loop current.) The **signal immediate** and **auto-cut-through** commands disable E&M signaling on this voice port. A G.711 audio packet stream is generated by the digital signal processor (DSP) on the E&M port.

If you are using an FXO voice port for live-feed audio stream instead of an E&M port, connect the source to the FXO voice port. This connection requires an external adapter to supply normal telephone company (telco) battery voltage with the correct polarity to the tip-and-ring leads of the FXO port. The adapter must also provide transformer-based isolation between the external audio source and the tip-and-ring leads of the FXO port.

If the **out-call** keyword is used, an outbound call to the live-feed source is attempted (or reattempted) every 30 seconds until the call is connected to the ephone-dn (extension) for which the **feed** command was configured. Note that this ephone-dn is not associated with a physical phone.

The related **moh** (ephone-dn) and **multicast moh** commands provide the ability to multicast an audio stream that is also being used as the source for Cisco CME system music on hold (MOH).

Note IP phones do not support multicast at 224.x.x.x addresses.

Examples

The following example sets up a call to extension 7777 for a live audio stream and sends it via multicast:

```
Router(config)# ephone-dn 55
Router(config-ephone-dn)# feed ip 239.1.1.1 port 2000 route 10.10.23.3 out-call 7777
```

Related Commands

Command	Description
auto-cut-through	Enables call completion when an M-lead response is not provided.
ip source-address	Identifies the IP address and port through which IP phones communicate with a Cisco CME router.
moh (ephone-dn)	Enables music on hold from a live feed and multicast of the MOH audio stream.
moh (telephony-service)	Enables music on hold from an audio file.
multicast moh	Enables multicast of a music-on-hold audio stream.
signal	Specifies the type of signaling for a voice port.

file text (voice register global)

To generate ASCII text files of the configuration profiles for SIP phones, use the **file text** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

file text
no file text

Syntax Description

This command has no arguments or keywords.

Command Default

System directly generates only binary files for configuration profiles.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

Use this command to generate an ASCII text fils of the configuration profile for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186s, or Cisco ATA-188s.

Examples

The following example shows how to generate an ASCII text file version of the configuration profiles for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186s, or Cisco ATA-188s:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# file text
Router(config-register-global)# create profile
```

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phone.
show voice register profile	Displays the contents of configuration files that are in ASCII text format.

filename

To specify a custom XML file that contains the dial patterns to use for a SIP dial plan, use the **filename** command in voice register dialplan configuration mode. To remove the file, use the **no** form of this command.

filename *filename*
no filename

Syntax Description

filename	Name of the XML file in flash memory.

Command Default

A custom file is not used for the dial plan.

Command Modes

Voice register dialplan configuration (config-register-dialplan)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command selects a custom XML file containing dial patterns for the SIP dial plan. The file specified with this command must be loaded into flash memory. You must use the **type** command to specify the type of phone for which the dial plan is being defined before you can use this command. After you define a dial plan, assign it to a SIP phone by using the **dialplan** command.

The **pattern** command and **filename** command are mutually exclusive. You can use either the **pattern** command to define dial patterns manually, or the **filename** command to select a custom dial pattern file that is loaded in system flash.

If the custom XML file contains any errors, the dial plan might not work properly on the phone.

To remove a dial plan that is created using a custom XML file, use the **reset** command after removing the dial plan from the phone and creating a new configuration profile. Removing a dial plan that uses a dial pattern XML file does not take effect if you restart the phone with the **restart** command.

Note This command is not supported for Cisco Unified IP Phone 7905 or 7912.

Examples

The following example shows that a custom file named sample.xml is specified for dial plan 2.

```
Router(config)# voice register dialplan 2
Router(config-register-dialplan)# type 7940-7960-others
Router(config-register-dialplan)# filename sample.xml
```

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.

Command	Description
pattern	Defines a dial pattern for a SIP dial plan.
show voice register dialplan	Displays all configuration information for a specific SIP dial plan.
type (voice register dialplan)	Defines a phone type for a SIP dial plan.

final

To define the last extension (ephone-dn) in an ephone hunt group, use the **final** command in ephone-hunt configuration mode. To remove this number from the hunt group, use the **no** form of this command.

final *number*
no final

Syntax Description

number	Extension or phone number. Can be an ephone-dn primary or secondary number, voice-mail number, pilot number of another hunt group, or FXS caller-ID number.

Command Default

No final number is defined.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command defines the last extension in a hunt group and the destination of incoming calls to a hunt-group pilot number that are unanswered after being routed through the directory numbers in the hunt group list.

To avoid an infinite loop, use the **max-redirect** command.

Examples

The following example defines ephone-dn 6000 as the last number of hunt group number 1:

```
Router(config)# ephone-hunt 1 sequential
Router(config-ephone-hunt)# final 6000
```

Related Commands

Command	Description
fwd-final	Specifies the final destination of an unanswered call that has been transferred into a hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the number of allowable redirects in a Cisco Unified CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of an ephone hunt group should not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.

Command	Description
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the ephone-hunt-group list.

final (voice hunt-group)

To define the last extension in a voice hunt group, use the **final** command in voice hunt-group configuration mode. To remove this number from the hunt group, use the **no** form of this command.

final *number*
no final

Syntax Description

number	Telephone or extension number. Can be an E.164 number, voice-mail number, pilot number of another hunt group, or FXS caller-ID number.

Command Default

No final number is defined in the voice hunt group.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command defines the last extension in a hunt group and the destination of incoming calls to a hunt-group pilot number that are unanswered after being routed through the directory numbers in the hunt group list.

To avoid an infinite loop, if a final number in one hunt group is configured as a pilot number of another hunt group, the pilot number of the first hunt group cannot be configured as a final number in any hunt group.

Examples

The following example shows how to define extension 6000 as the last number of hunt group 1:

```
Router(config)# voice hunt-group 1 sequential
Router(config-voice-hunt-group)# final 6000
```

Related Commands

Command	Description
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next number in a peer hunt-group list before proceeding to the final number.
list (voice hunt-group)	Defines the numbers that participate in a voice hunt group.
max-redirect (voice register global)	Changes the current number of allowable redirects in a Cisco CME system.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

forward local-calls

To allow internal (local) calls to be forwarded, use the **forward local-calls** command in ephone-dn or ephone-hunt configuration mode. To prevent internal calls from being forwarded, use the **no** form of this command.

forward local-calls
no forward local-calls

Syntax Description

This command has no arguments or keywords.

Command Default

Internal calls are forwarded as specified in the ephone-dn or ephone-hunt configuration of the called party.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Internal, or local, calls are defined as those calls that originate from other ephone-dns in the same Cisco Unified CME system.

When the **no forward local-calls** command is used in ephone-dn configuration mode, internal calls to that ephone-dn are not forwarded if the ephone-dn is busy or does not answer. If the ephone-dn is busy, the caller hears a busy signal.If the ephone-dn does not answer, the caller hears a ringback signal. The call is not forwarded even if call forwarding is enabled for the ephone-dn.

When the **no forward local-calls** command is used in ephone-hunt configuration mode, internal calls to a hunt-group pilot number are sent only to the first member of the group. If the first group member is busy, the caller hears a busy signal. If the first group member does not answer, the caller hears a ringback signal. The call is not forwarded to subsequent hunt group members.

Examples

In the following example, extension 2222 dials the pilot number 3000 and is forwarded to extension 3011. If 3011 is busy, the caller hears a busy tone. If 3011 does not answer, the caller hears ringback. The call is not forwarded, even after the timeout expires.

```
ephone-hunt 17 sequential
 pilot 3000
 list 3011, 3021, 3031
 timeout 10
 final 7600
 no forward local-calls
```

In the following example, extension 2222 calls extension 3675 and hears ringback or a busy signal. If an external caller reaches extension 3675 and there is no answer, the call is forwarded to extension 4000.

```
ephone-dn 25
 number 3675
 no forward local-calls
 call-forward noan 4000 timeout 30
```

forward local-calls (voice hunt-group)

To allow local calls to be forwarded, use the **forward local-calls** command in voice hunt-group configuration mode. To prevent local calls from being forwarded, use the **no** form of this command.

forward local-calls to-final
no forward local-calls to-final

Syntax Description

to-final	Prevents local calls from being forwarded to the final destination number.

Command Default

Local calls are forwarded as specified in the voice hunt-group configuration of the called party.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

Local or internal calls are calls originating from a Cisco Unified SIP or Cisco Unified SCCP IP phone in the same Cisco Unified CME system.

Before Cisco Unified CME 9.5, the **no forward local-calls** command was configured in ephone-hunt group to prevent a local call from being forwarded to the next agent.

In Cisco Unified CME 9.5, local calls are prevented from being forwarded to the final destination using the **no forward local-calls to-final** command in parallel or sequential voice hunt-group configuration mode.

When the **no forward local-calls to-final** command is configured in sequential voice hunt-group configuration mode, local calls to the hunt-group pilot number are sent sequentially only to the list of members of the group using the rotary-hunt technique. In case all the group members of the voice hunt group are busy, the caller hears a busy tone. If any of the group members are available but do not answer, the caller hears a ringback tone and is eventually disconnected after the specified timeout. The call is not forwarded to the final destination number.

When the **no forward local-calls to-final** command is configured in parallel voice hunt-group configuration mode, local calls to the hunt-group pilot number are sent parallely to the list of members of the group using the blast technique. In case all the group members of the voice hunt group are busy, the caller hears a busy tone. If any of the group members are available but do not answer, the caller hears a ringback tone and is eventually disconnected after the specified timeout.The call is not forwarded to the final destination number.

Examples

The following example shows how to prevent the forwarding of local calls to the final destination in parallel voice hunt group 1:

```
Router# configure terminal
Router(config)# voice hunt-group 1 parallel
Router(config-voice-hunt-group)# no forward local-calls to-final
```

Related Commands

Command	Description
voice hunt-group	Enters voice hunt-group configuration mode to create a hunt group for phones in a Cisco Unified CME system.

forwarding local (voice register global)

To use the forwarding-party number and name (the local number and name) in calls forwarded using local hairpin call routing on a SIP phone, use the **forwarding local** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

forwarding local
no forwarding local

Syntax Description

This command has no arguments or keywords.

Command Default

Calling-party name and number used.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command replaces a calling-party number and name with the local forwarding-party number and name in hairpinned forwarded calls.

Examples

The following example shows how to enable local forwarding:

```
Router(config)# voice register global
Router(config-register-global)# forwarding local
```

Related Commands

Command	Description
call-forward b2bua all (voice register dn and voice register pool)	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.

from-ring

To specify that on-hook time stamps for ephone hunt group agents should be updated when calls ring as well as when calls are answered in a longest-idle ephone hunt group, use the **from-ring** command in ephone-hunt configuration mode. To return to the default, use the **no** form of this command.

from-ring
no from-ring

Syntax Description

This command has no keywords or arguments.

Command Default

On-hook time stamps are updated only when calls are answered by agents.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used only with longest-idle ephone hunt groups. In a longest-idle hunt group, the algorithm for choosing the the next agent to receive a call is based on a comparison of on-hook time stamps. The agent with the smallest on-hook time stamp value is chosen when the next call comes to the hunt group.

This command can be used to specify that on-hook time stamps should be updated when calls ring agents as well as when calls are answered by agents.

The **show ephone-hunt** command displays on-hook time stamps.

Examples

The following example defines longest-idle ephone hunt group 1 with a pilot number 7501, a final number 8000, and five numbers in the list. Because the **from-ring** command is used, on-hook time stamps will be recorded when calls ring agents as well as when calls are answered. After a call is redirected three times (makes six hops), it is redirected to the final number, 8000.

```
ephone-hunt 1 longest-idle
 pilot 7501
 list 7001, 7002, 7023, 7028, 7045
 final 8000
 from-ring
 hops 3
 timeout 20
telephony-service
 max-redirect 8
```

Related Commands

Command	Description
show ephone-hunt	Displays configuration information, current status, and statistics for ephone hunt groups.

fwd-final

To specify the final destination of a call that has been transferred into a hunt group and is unanswered, use the **fwd-final** command in ephone-hunt configuration mode. To return to the default, use the **no** form of this command.

fwd-final {**orig-phone** | **final**}
no fwd-final {**orig-phone** | **final**}

Syntax Description

orig-phone	Phone that originally answered a call before transferring it to the pilot number of a hunt group.
final	Last extension in the hunt group as specified in the hunt group configuration.

Command Default

Calls are sent to the final number that is specified in the hunt group configuration.

Command Modes

Ephone-hunt configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used for routing only unanswered transferred calls. Transferred calls are incoming calls to an ephone hunt group that were previously answered by a Cisco Unified CME extension and transferred into the hunt group.

The **orig-phone** keyword specifies that an unanswered transferred call is routed back to the extension that originally answered the call and transferred it to the hunt group.

The **final** keyword specifies that an unanswered transferred call is routed to the last extension in the hunt group as defined by using the **final** command.

Examples

The following example sets up a peer hunt group with three ephone-dns to answer calls. An unanswered transferred call will be routed to the ephone-dn that transferred it to the ephone hunt group pilot number. A DID call that dials the pilot number directly will be routed to extension 7600 if it is unanswered by the hunt group.

```
ephone-hunt 17 peer
 pilot 3000
 list 3011, 3021, 3031
 hops 3
 final 7600
 fwd-final orig-phone
```

Related Commands

Command	Description
final	Defines the last extension (ephone-dn) in an ephone hunt group.

fxo hook-flash

To enable display of a flash soft key on a Cisco IP Phones 7940 and 7940G or Cisco IP Phones 7960 and 7960G in a Cisco CallManager Express (Cisco CME) system, use the **fxo hook-flash** command in telephony-service configuration mode. To disable display of the flash soft key, use the **no** form of this command.

fxo hook-flash
no fxo hook-flash

Syntax Description

This command has no arguments or keywords.

Command Default

The flash soft key is disabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Certain public switched telephony network (PSTN) services, such as three-way calling and call waiting, require hookflash intervention from the phone user. A soft key labeled flash provides this functionality for the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G users on foreign exchange office (FXO) lines attached to the Cisco CME system. The flash soft key is enabled using the **fxo hook-flash** command.

Once a flash soft key has been enabled on an IP phone, it is available to provide hookflash functionality during all calls except local IP-phone-to-IP-phone calls. Note that hookflash-controlled services can be activated only if they are supported by the PSTN connection that is involved in the call. The availability of the flash soft key does not guarantee that hookflash-based services are actually accessible to the phone user.

The flash soft key display is automatically disabled for local IP-phone-to-IP-phone calls.

This command must be followed by a quick reboot of the phones using the **restart all** command.

Examples

The following example enables the flash soft key on the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G:

```
Router(config)# telephony-service
Router(config-telephony)# fxo hook-flash
```

Related Commands

Command	Description
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

Cisco Unified CME Commands: G

gsm-support

To define the gsm support for a Cisco Unified SIP IP phone on Cisco Unified CME, use the **gsm-support** command in voice register pool-type mode. To remove the gsm support, use the **no** form of this command. no form is typically used to override the inherited property of the reference ephone with default value.

gsm-support
nogsm-support

Syntax Description

This command has no arguments or keywords.

Command Default

By default, the gsm-support is not enabled. When **reference-pooltype** is configured, the **gsm-support** value of the reference phone is inherited.

Command Modes

Voice Register Pool-type Configuration (config-register-pooltype)

Command History

Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to define the maximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME. When you use the **no** form of this command, the inherited properties of the reference phone is takes precedence over the default value.

Cisco Unified CME

The following example shows how to enter voice register pool configuration mode and define themaximum number of addon modules for a Cisco Unified SIP IP phone on Cisco Unified CME:

```
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# gsm-support
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.

group (lpcor custom)

To add a logical partitioning class of restriction (LPCOR) resource group to the custom resource list, use the **group** command in LPCOR custom configuration mode. To remove a resource group, use the **no** form of this command.

group *number* *lpcor-group*
no group *number*

Syntax Description

number	Group number of the LPCOR entry. Range: 1 to 64.
lpcor-group	Name of a LPCOR resource group.

Command Default

LPCOR resource group is not defined.

Command Modes

LPCOR custom configuration (cfg-lpcor-custom)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use this command to define all of the LPCOR resource groups that you are provisioning on the Cisco Unified CME router. You must logically partition the resources of the Cisco Unified CME router (trunks and phones) into different LPCOR resource groups so that you can apply the required call restrictions to each group.

Examples

The following example shows a LPCOR configuration with six resource groups:

```
voice lpcor custom
 group 1 sccp_phone_local
 group 2 sip_phone_local
 group 3 analog_phone_local
 group 4 sip_remote
 group 5 sccp_remote
 group 6 isdn_local
```

Related Commands

Command	Description
voice lpcor enable	Enables LPCOR functionality on the Cisco Unified CME router.
voice lpcor policy	Creates a LPCOR policy for a resource group.

group (telephony-service)

To create a (VRF) group for Cisco Unified CME users and phones, use the **group** command in telephony-service configuration mode. To remove a group, use the **no** form of this command.

group *group-tag* [**vrf** *vrfname*]
no group

Syntax Description

group-tag	Unique identifier for VRF group being configured. Range 1 to 5
vrf *vrfname*	(Optional) Name of already-configured VRF to which this VRF group is associated.

Command Default

No group is configured.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Release	Modification
12.4(22)T	This command was introduced.

Usage Guidelines

By default, VRF groups are associated with a global voice VRF unless you use the **vrf** *vrfname* keyword and argument combination to specifiy otherwise.

If you configure this command, the **ip source-address**, **url** and **cnf-file location** commands in **telephony-service** configuration mode are automatically converted into *group 1* with a default global VRF for nvgen during system upgrade.

If you configure this command and the **cnf-file location** command is configured for **system:**, the per phone or per phone type file for an ephone in the VRF group is created in *system:/its/vrf<group-tag>/* . Local files are still created in system:/its/.

If you configure this command and the **cnf-file location** command is configured as **flash:** or **slot0:**, the per phone or per phone type file for an ephone in the VRF group is named *flash:/its/vrf<group-tag>_<filename> or slot0:/its/vrf<group tag>_filename>* .

The location of the locale files is not affected by configuring a VRF group.

Examples

The following example shows the configuration for three VRF groups. Group 1 is on a global voice VRF and the other two groups are on data VRFs.

```
telephony-service
 sdspfarm conference mute-on # mute-off #
 sdspfarm units 4
 sdspfarm transcode sessions 10
 sdspfarm tag 1 xcode101
 sdspfarm tag 2 conf103
 group  1
  ip source-address 10.1.10.1 port 2000
  url directories http://210.1.10.1/localdirectory
 !
 group  2 vrf data-vrf1
```

```
  ip source-address 10.2.10.1 port 2000
 !
 group  3 vrf data-vrf2
  ip source-address 10.3.10.1 port 2000
 !
.
.
.
```

Related Commands

Command	Description
ip vrf	Defines a VRF for a router.
cnf-file location	Specifies a storage location for phone configuration files

group phone

To add a phone, including a TAPI-based client application, or a softphone on a PC to a VRF group for Cisco Unified CME, use the **group** command in ephone or ephone-template configuration mode. To remove the **group** configuration, use the **no** form of this command.

group
no group

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this feature is disabled.

Command Modes

Ephone configuration (config-ephone)

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Products	Modification
12.4(22)T	Cisco Unified CME 7.0(1)	This command was introduced.

Usage Guidelines

This command enables to configure the voice VRF group for SIP phones. This command adds a softphone on a PC, an IP phone, or a TAPI client on an IP phone to a VRF group.

VRF groups for users and phones in Cisco Unified CME are created by using the **group** command in telephony-service configuration mode. All SCCP and SIP phones connected to Cisco Unified CME must register through the global voice VRF. TAPI-based client on an IP phone and softphones on a PC must register in Cisco Unified CME through a data VRF.

Before you can use this command, the MAC address for the IP phone being configured must be configured by using the **mac-address** command in ephone configuration mode.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode, the value that you set in ephone configuration mode has priority over the ephone-template configuration.

Examples

The following example shows four phones in three VRF groups, two on data VRFs and one on a global voice VRF.

```
telephony-service
 sdspfarm conference mute-on # mute-off #
 sdspfarm units 4
 sdspfarm transcode sessions 10
 sdspfarm tag 1 xcode101
 sdspfarm tag 2 conf103
 group  1
  ip source-address 209.165.201.1 port 2000
  url directories http://209.165.201.1/localdirectory
 !
 group  2 vrf data-vrf1
  ip source-address 209.165.201.2 port 2000
 !
```

```
group  3 vrf data-vrf2
  ip source-address 209.165.201.3 port 2000
 !
.
.
!
ephone-template 1
  group phone 1 tapi 2
ephone-template 2
  group phone 2
...
ephone 1
  mac-address 1111.2222.3333
  ephone-template 1
ephone 2
  mac-address 2222.2222.3333
  ephone-template 2
ephone 3
  mac-address 1111.3333.3333
  group phone 1 tapi 3
ephone 4
  mac-address 1111.2222.4444
  group phone 3
!
```

The following example shows four phones in three VRF groups, two on data VRFs and one on a global voice VRF.

```
Router(config)# voice register template
Router(config-telephony)# group <group-tag>
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode.
voice register template	Enters voice register template configuration mode.
ephone-template (ephone)	Applies an ephone template to an ephone configuration.
group (telephony-service)	Creates a VRF group for phones and users in Cisco Unified CME.
mac-address	Associates the MAC address of a Cisco IP phone with an ephone configuration.

group (voice register global)

To add a phone or a softphone on a PC to a Virtual routing and forwarding (VRF) group for Cisco Unified CME, use the **group** command in voice register global configuration mode. To remove the configuration, use the **no** form of this command. To configure SIP CME support for VRF by provisioning its source address under a group, use the **vrfname** command. To remove the configuration, use the **no** form of this command.

group *group-tag*
no group
group *group-tag* **vrf***vrfname*
no group vrf*vrfname*

Syntax Description

group-tag	Unique identifier of VRF group. Range is from 1 to 5.
vrfname	Specifies the name of the vrf group.

Command Default

By default, this feature is disabled.

Command Modes

voice register global (config-register global)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

This command enables to configure the VRF group for SIP phones. This command is used to configure multiple VRF groups for SIP phones and soft phones on PC or mobile devices registering to CME. Each VRF group can be associated with a specific IP VRF. Phones in this VRF will use the source-address configured under this VRF group to register to CME.

Example

The following example shows three different VRF groups that have been configured, a voice VRF, a Data VRF, and a Voice VRF:

```
vovoice register global
 mode  cme
 max-dn 100
 max-pool 100

 group 1 vrf voice-vrf
  source-address 8.0.0.1

 group 2 vrf data-vrf
  source-address 9.0.0.1

 group 3 vrf voice-vrf1
  source-address 10.0.0.1
```

Related Commands

Command	Description
group (telephony-service)	Creates a VRF group for phones and users in Cisco Unified CME.

group (voice register pool)

To configure multiple virtual routing and forwarding (VRF) groups for SIP phones and soft phones on PC or mobile, use the **group** command in voice register pool configuration or voice register template configuration modes. To remove the configuration, use the **no** form of this command. You can configure upto 5 VRF groups.

To add a phone or a softphone on a PC to a VRF group for Cisco Unified CME, use the **group** command in voice register pool or voice register template configuration modes. To remove the configuration, use the **no** form of this command.

group *group-tag*
no group

Syntax Description

group-tag	Unique identifier of VRF group. Range is from 1 to 5.

Command Default

By default, this feature is disabled.

Command Modes

voice register pool (config-register pool)

voice register template (config-register template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

This command enables to configure the VRF group for SIP phones. The group id configured under voice register global can be associated to the voice register pool or template using the **group <group-tag>** command.

Example

The following example shows phones configured in different VRF groups under voice register pool and voice register template modes:

```
voice register pool  1
group 1

voice register template 1
group 3
```

Related Commands

Command	Description
voice register global	Enters voice register global configuration mode.
voice register template	Enters voice register template configuration mode.

Cisco Unified CME Commands: H

headset auto-answer line

To enable auto-answer on the specified line when the headset key is engaged, use the **headset auto-answer** command in ephone configuration mode. To disable headset auto-answer for this line, use the **no** form of this command.

headset auto-answer line line-number
no headset auto-answer line line-number

Syntax Description

line-number	Phone line that should be automatically answered.

Command Default

Headset auto-answer is not enabled.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command enables headset auto-answer on a particular line. A line, as used in this command, is *not* identical to a phone button. A line, as used in this command, represents the ability for a call connection on this phone, and the line numbers generally follow a top-to-bottom sequence starting with the number 1.

The following examples represent common situations pertaining to a button:line relationship:

- button 1:1—A single ephone-dn is associated with a single ephone button. Counts as one line.
- button 1o1,2,3,4,5—Five ephone-dns are overlaid on a single ephone button. Counts as one line.
- button 2x1—An ephone button acts as an extension for an overlaid ephone button. Counts as one line.
- Button is unoccupied or programmed for speed-dial. Does not count as a line.

Examples

The following example shows how to enable headset auto-answer for line 1 (button 1) and line 4 (button 4), which has overlaid ephone-dns but counts as a single line in this context. In this example, four (1, 2, 3, and 4) buttons are defined for ephone 3.

```
ephone 3
 button 1:2 2:4 3:6 4o21,22,23,24,25
 headset auto-answer line 1
 headset auto-answer line 4
```

The following example shows how to enable headset auto-answer for line 2 (button 2), which has overlaid ephone-dns, and line 3 (button 3), which is an overlay rollover line. In this example, three (1, 2, and 3) buttons are defined for ephone 17.

```
ephone 17
 button 1:2 2o21,22,23,24,25 3x2
 headset auto-answer line 2
```

```
 headset auto-answer line 3
```

The following example shows how to enable headset auto-answer for line 2 (button 3) and line 3 (button 5). In this case, the button numbers do not match the line numbers because buttons 2 and 4 are not used.

```
ephone 25
 button 1:2 3:4 5:6
 headset auto-answer line 2
 headset auto-answer line 3
```

hfs enable

To enable the HTTP File-Fetch Server (HFS) download service on an IP Phone in a Cisco Unified CME system, use the **hfs enable** command in telephony-service configuration mode. To disable the HFS download service, use the **no** form of this command.

hfs enable [**port** *port-number*]
no hfs enable [**port** *port-number*]

Syntax Description

port *port-number*	(Optional) Specifies the port where the HFS download service is enabled. **Range is from 1024 to 65535.** **Note** If the entered custom HFS port clashes with the underlying IP HTTP port, an error message is displayed and the command is disallowed.

Command Default

An IP Phone is unable to download configuration and firmware files through the HFS infrastructure.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

To enable the HFS download service, the underlying HTTP server must be enabled first using the **ip http server** command because the HFS infrastructure is built on top of an existing IOS HTTP server.

This HFS infrastructure enables multiple HTTP services to co-exist. The HFS download service runs on custom port 6970 but can also share default port 80 with other services. Other HTTP services run on other non-standard ports like 1234.

Use the **hfs enable** command without keyword or argument to enable the HFS download service on the default HTTP server port.

Examples

The following example shows how to enable the HFS download service for Cisco Unified SIP IP Phone 7945 on port 65500:

```
Router(config)# ip http server
Router(config)# ip http port 1234
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# load 7945 SIP45.8.3.3S
Router(config-register-global)# create profile
Router(config-register-global)# exit
Router (config)# telephony-service
Router(config-telephony)# hfs enable port 65500
```

The following examples show how to enable the HFS service on default and custom ports.

For the default port:

```
Router(config)# ip http server
```

```
Router(config)# ip http port 1234
.
.
Router (config)# telephony-service
Router(config-telephony)# hfs enable
```

For the custom port:

```
Router(config)# ip http server
Router(config)# ip http port 1234
.
.
Router (config)# telephony-service
Router(config-telephony)# hfs enable port 6970
```

The following example shows how an entered custom HFS port clashes with the underlying ip http port. Port 6970 is configured as the IP HTTP port. When the HFS port is configured with the same value, an error message is displayed to show that the port is already in use.

```
Router(config)# ip http server
Router (config)# ip http port 6970
.
.
Router (config)# telephony-service
Router (config-telephony)# hfs enable port 6970
.
Invalid port number or port in use by other application
```

The HFS port number is already in use by the underlying IP HTTP server so an HFS port that is different from the underlying IP HTTP port must be used.

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profile files required for SIP phones.
ip http port	Specifies the port where the HTTP service is run.
ip http server	Enables the underlying IOS HTTP server of the the HFS infrastructure.

hfs home-path

To set up a home-path for IP phone firmware files, use the **hfs home-path** command in telephony-service configuration mode. To remove a directory as a home-path for phone files, use the **no** form of this command.

hfs home-path *path*
no hfs home-path *path*

Syntax Description

path	Directory path where only IP phone firmware and configuration files are stored. **Note** The administrator must store the phone firmware files at the location set as the home path directory

Command Default

No directory path is specified for the storage of IP phone firmware and configuration files.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

Use the **hfs home-path** command to specify a directory path as the home-path to store IP phone firmware files.

Examples

The following example shows how to set up a home-path for IP phone firmware files in Cisco Unified CME:

```
Router(config)# telephony service
Router(config-telephony)# hfs home-path flash:/cme/loads/
```

The following example shows how a new directory called phone-load can be created under the root directory of the flash memory and set as the hfs home-path:

```
cassini-c2801#mkdir flash:phone-loads
Create directory filename [phone-loads]?
Created dir flash:phone-loads
cassini-c2801#sh flash:
-#- --length-- -----date/time------ path
1     13932728 Mar 22 2007 15:57:38 +00:00 c2801-ipbase-mz.124-1c.bin
2     33510140 Sep 18 2010 01:21:56 +00:00 rootfs9951.9-0-3.sebn
3       143604 Sep 18 2010 01:22:20 +00:00 sboot9951.111909R1-9-0-3.sebn
4         1249 Sep 18 2010 01:22:40 +00:00 sip9951.9-0-3.loads
5        66996 Sep 18 2010 01:23:00 +00:00 skern9951.022809R2-9-0-3.sebn
6        10724 Sep 18 2010 00:59:48 +00:00 dkern9951.100609R2-9-0-3.sebn
7      1507064 Sep 18 2010 01:00:24 +00:00 kern9951.9-0-3.sebn
8            0 Jan 5 2011 02:03:46 +00:00 phone-loads
14819328 bytes available (49192960 bytes used)
cassini-c2801#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
cassini-c2801(config)#tele
cassini-c2801(config)#telephony-service
cassini-c2801(config-telephony)#hfs hom
```

```
cassini-c2801(config-telephony)#hfs home-path flash:?
WORD
cassini-c2801(config-telephony)#hfs home-path flash:phone-loads
cassini-c2801(config-telephony)#
```

Related Commands

Command	Description
hfs enable	Enables the HFS download service on an IP Phone in a Cisco Unified CME system.

hlog-block (voice hunt-group)

To disable agent status control (logout/login) for voice hunt group on SIP or SCCP phones using Hlog softkey or by using FAC, use the **hlog-block** command in voice hunt-group configuration mode. To remove the configuration, use the **no** form of this command.

hlog-block
no hlog-block

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this command is disabled.

Command Modes

voice hunt group configuration (config-voice-hunt-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M5	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

This command disables the agent status control such that SIP or SCCP phones are not be able to logout/login from the voice hunt group using Hlog/FAC.

Examples

The following example shows how the voice hunt group hlog-block option is enabled for a phone:

```
Router(config)# voice hunt-group 1 parallel
Router(config-voice-hunt-group)# hlog-block
```

hold-alert

To set a repeating audible alert notification when a call is on hold on a Cisco Unified IP phone, use the **hold-alert** command in ephone-dn or ephone-dn-template configuration mode. To disable this feature, use the **no** form of this command.

hold-alert *timeout* {**idle** | **originator** | **shared** | **shared-idle**} [**recurrence** *recurrence-timeout*] [**ring-silent-dn**]
no hold-alert *timeout* {**idle** | **originator** | **shared** | **shared-idle**} [**recurrence** *recurrence-timeout*] [**ring-silent-dn**]

timeout	Interval after which an audible alert notification is repeated, in seconds. Range is from 15 to 300. There is no default.
idle	Alerts only when the phone is idle.
originator	Alerts whether the phone is idle or busy.
shared	Alerts only when the extension is idle but alerts all phones that share the line.
shared-idle	Alerts all idle phones that share the line.
recurrence	Alerts recurrence after first timeout.
recurrence-timeout	Call on-hold recurrence timeout in seconds. Range is from 2 to 300.
ring-silent-dn	Rings the silent DN.

Command Default

Audible alert notification for on-hold calls is disabled. Only a visual indication is provided.

Command Modes

Ephone-dn configuration (config-ephone)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T .
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode. The shared-idle option and ring-silent-dn parameter were introduced.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.
15.1(3)T2	Cisco Unified CME 8.5	The recurrence parameter was introduced.
15.1(4)M1	Cisco Unified CME 8.6	The recurrence parameter was introduced.

Usage Guidelines

Use the **hold-alert** command to set an audible alert notification on a Cisco Unified IP phone to remind the phone user that a call is on hold. The *timeout* argument specifies the time interval in seconds from the time the call is placed on hold to the time the on-hold audible alert is generated. The alert is repeated every *timeout* seconds.

When the **idle** keyword is enabled, a one-second burst of ringing on the phone is generated on the IP phone that placed the call in the hold state, but only if the phone is in the idle state. If the phone is in active use, no on-hold alert is generated.

When the **originator** keyword is enabled, a one-second burst of ringing is generated on the phone that placed the call in the hold state, but only if the phone is in the idle state. If the phone is in use on another call, an audible beep (call-waiting tone) is generated.

When the **shared** keyword is enabled, a one-second ring burst is generated for all the idle phones that share the extension with the on-hold call. Phones that are in use do not receive an audio beep (call-waiting tone) alert. Only the phone that placed the call on hold hears a call-waiting beep if it is busy.

When the **shared-idle** keyword is enabled, a one-second ring burst is generated for all the idle phones that share the line with the on-hold call.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example sets audible alert notification to idle on extension 1111:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1111
Router(config-ephone-dn)# name phone1
Router(config-ephone-dn)# hold-alert 100 idle
```

The following example uses an ephone-dn template to set audible alert notification for extension 1111 to only occur when the phone is idle:

```
Router(config)# ephone-dn-template 3
Router(config-ephone-dn-template)# hold-alert 100 idle
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1111
Router(config-ephone-dn)# name phone1
Router(config-ephone-dn)# ephone-dn-template 3
The following example uses an ephone-dn to set an additional timeout value between 2 and
300.
Router(config-ephone-dn)# hold-alert
  <15-300> call on-hold timeout in seconds
Router(config-ephone-dn)# hold-alert 15
  idle         alert on-hold originator only if idle
  originator   alert on-hold originator always
  shared       alert all phones that share the line
  shared-idle  alert all idle phones that share the line
Router(config-ephone-dn)# hold-alert 15 idle
  recurrence      alternate alert recurrence timeout after first
  ring-silent-dn  ring the silent DN
Router(config-ephone-dn)# hold-alert 15 idle recurrence
  <2-300>  call on-hold recurrence timeout in seconds
Router(config-ephone-dn)# hold-alert 15 idle recurrence 3
  ring-silent-dn  ring the silent DN
```

The following example uses an ephone-dn-template to set an additional timeout value between 2 and 300.

```
Router(config-ephone-dn-template)# hold-alert
  <15-300> call on-hold timeout in seconds
Router(config-ephone-dn-template)# hold-alert 15
  idle         alert on-hold originator only if idle
  originator   alert on-hold originator always
  shared       alert all phones that share the line
  shared-idle  alert all idle phones that share the line
Router(config-ephone-dn-template)# hold-alert 15 idle
  recurrence      alternate alert recurrence timeout after first
  ring-silent-dn  ring the silent DN
Router(config-ephone-dn-template)# hold-alert 15 idle recurrence
  <2-300>  call on-hold recurrence timeout in seconds
Router(config-ephone-dn-template)# hold-alert 15 idle recurrence 3
  ring-silent-dn  ring the silent DN
```

Related Commands

Command	Description
ephone-dn-template (ephone-dn)	Applies ephone-dn-template to the ephone-dn being configured.

hold-alert (voice register global)

To enable a one-time audible alert notification for a call still on hold after a subsequent call has ended in Cisco Unified CME, use the command in voice register global configuration mode. To disable this feature, use the **no** form of this command.

hold-alert
no hold-alert

Syntax Description

This command has no arguments or keywords.

Command Default

Audible alert notification for on-hold calls is disabled.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command enables a one time audible alert notification on all supported SIP phones in a Cisco Unified CME system to remind the phone user that a call is on hold after a subsequent call has ended. The alert is not repeated and does not alert until a subsequent call ends after the original call was placed on hold. This applies globally to all SIP CME Phones.

Note This command does not apply to Cisco ATAs that have been configured for SIP in Cisco Unified CME.

Examples

The following example shows how to set audible alert notification on SIP phones for on-hold calls:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# hold-alert
Router(config-register-global)# create profile
! Restart phone(s) to update config file change
```

Related Commands

Command	Description
call-waiting (voice register pool)	Enables call waiting on a SIP phone.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.

hops

To define the number of times that a call can proceed to the next ephone-dn in a peer or longest-idle ephone hunt group before the call proceeds to the final ephone-dn, use the **hops** command in ephone hunt configuration mode. To return to the default number of hops, use the **no** form of this command.

hops *number*
no hops *number*

Syntax Description

number	Number of hops before the call proceeds to the final ephone-dn. Range is from 2 to 20, but the value must be less than or equal to the number of extensions that are specified in the **list** command. Default automatically adjusts to the number of hunt group members.

Command Default

The number of hops automatically adjusts to the number of ephone hunt group members.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command Modes

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The maximum number of hops was restricted to the number of extensions specified in the **list** command.
12.3(11)XL	Cisco CME 3.2.1	Increased maximum number of hops to 20.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The default was changed from 2 hops to automatically adjust the number of hops to the number of ephone hunt group members.
12.4(9)T	Cisco Unified CME 4.0	The modification to change the default was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is valid only for peer and longest-idle ephone hunt groups in Cisco Unified CallManager Express systems.

This command is required when you are configuring the **auto logout** command for peer and longest-idle hunt groups.

Examples

The following example sets the number of hops to 6 for peer hunt group 3:

```
Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# hops 6
```

Related Commands

Command	Description
auto logout	Enables the automatic change of an ephone hunt group agent's ephone-dn to not-ready status.
final	Defines the last ephone-dn in an ephone hunt group.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco Unified CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

hops (voice hunt-group)

To define the number of times that a call can hop to the next number in a peer hunt group before the call proceeds to the final number, use the **hops** command in voice hunt-group configuration mode. To return to the default number of hops, use the **no** form of this command.

hops *number*
no hops

Syntax Description

number	Number of hops before the call proceeds to the final number. Range is 2 to 10, but the value must be less than or equal to the number of extensions that are specified in the **list** command. The default is the same number as there are destinations defined under the **list** command.

Command Default

The default is the number of *directory-number* arguments configured in the **list** command.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Cisco IOS Release	Cisco product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command is valid only for peer or longest-idle voice hunt groups in Cisco Unified CME systems.

Examples

The following example shows how to set the number of hops to 6 for peer voice hunt group 1:

```
Router(config)# voice hunt-group 1 peer
Router(config-voice-hunt-group)# list 1000, 1001, 1002, 1003, 1004, 1005, 1006, 006, 1007,
 1008, 1009
Router(config-voice-hunt-group)# hops 6
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
list (voice hunt-group)	Defines the directory numbers that participate in a hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last directory number in the hunt group.

host-id-check

To configure host-id-check option for a vpn-profile, use the **host-id-check** command in vpn-profile configuration mode. To disable the host-id-check configuration, use the no form of this command.

host-id-check [{*enabledisable*}]

Syntax Description

enable	Enables host-id-check option in a vpn-profile.
disable	Disables host-id-check option in a vpn-profile.

Command Default

Host-id-check option is enabled.

Command Modes

Vpn-profile configuration (conf-vpn-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to configure host-id-check option for a vpn-profile. This host ID check enhances the security by parsing the host name or the IP from latest URL of the VPN concentrator to check against the subjectAltNames field within the certificate, if the subjectAltNames existed. This check is performed by the phone.

Examples

The following example shows the host-id-check option enabled in vpn-profile 2 and disabled in vpn-profile 1:

```
Router# show run
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  keepalive 50
  host-id-check disable
 vpn-profile 2
  mtu 1300
  password-persistent enable
  host-id-check enable
 sip
!
voice class media 10
 media flow-around
!
!
voice register global
 max-pool 10
```

Related Commands

Command	Description
vpn-profile	Defines a VPN-profile.

hunt-group report url

To set the filename parameters and the URL path where hunt group call statistics are sent using TFTP, use the **hunt-group report url** command in telephony service mode. To disable this feature, use the **no** form of this command.

hunt-group report url {**prefix** | **suffix**}
no hunt-group report url {**prefix** | **suffix**}

prefix	Provides the prefix of the filename to which the statistics are written. The prefix of the filename appears at the end of the URL.
suffix	Provides the suffix of the filename to which the statistics are written. Range is <0-1> to <1-200>.

Command Default

This command is disabled by default.

Command Modes

telephony-service (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T .

Related Commands

Command	Description
hunt-group report delay hours	Delays the automatic transfer of Cisco CME B-ACD call statistics to a file.
hunt-group report every hours	Sets the hourly interval after which Cisco CME B-ACD call statistics are automatically transferred to a file.

hunt-group statistics write-v2

To write all the ephone hunt and voice hunt group statistics to a file along with total logged in and logged out time for agents, use the **hunt-group statistics write-v2** command in privileged EXEC mode.

hunt-group statistics write-v2 *location*

Syntax Description

location	URL or filename to which the statistics is written.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(2)T	Cisco Unified CME 9.5	This command was introduced.
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was enhanced to add statistics for total logged in and logged out time for voice hunt group.

Usage Guidelines

Use the **hunt-group statistics write-v2** command to write out, in hourly increments, all the ephone and voice hunt group statistics for the past seven days, along with total logged in and logged out time for agents. This command is intended to be used when normal hunt group statistics collection is interrupted, perhaps due to TFTP server failure. If applicable, the TFTP statistics report consists of both ephone and voice hunt statistics.

Note On the day that daylight saving time adjusts the time back by one hour at 2 a.m. each year, the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

Examples

The following example shows how the **hunt-group statistics write-v2** command writes a combination of ephone and voice hunt group statistics to a file in TFTP server 202.153.144.25:

```
Router# hunt-group statistics write-v2 tftp://202.153.144.25/cmeteam/stats
Writing out all hunt group statistics to tftp://202.153.144.25/cmeteam/stats
01:47:08 UTC Mon Mar 21 2016,
,
EPHONE HUNT GROUP STAT,
01, Sat 00:00 - 01:00, HuntGp, 02, 02, 00001, 00001, 00000, 0007, 0007, 000001, 000001,
0000, 00000, 000000, 000000,
01, Sat 00:00 - 01:00, Agent, 5012, 00001, 000001, 000001, 00000, 000000, 000000, 00001,
000008, 000008, 00001, 000003, 000003,
01, Sat 00:00 - 01:00, Queue, 00000, 00001, 00000, 00006, 00006, 00000, 00000, 00000, 00000,
 00000,
01, Sat 01:00 - 02:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 02:00 - 03:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 03:00 - 04:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 04:00 - 05:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 05:00 - 06:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
```

```
0000, 00000, 000000, 000000,
01, Sat 06:00 - 07:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 07:00 - 08:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000,
.
.
.
06, Fri 19:00 - 20:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
06, Fri 20:00 - 21:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
06, Fri 21:00 - 22:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
06, Fri 22:00 - 23:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
06, Fri 23:00 - 00:00, HuntGp, 02, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
VOICE HUNT GROUP STAT,
01, Fri 01:00 - 02:00, HuntGp, 03, 02, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Fri 01:00 - 02:00, Agent, 2001, 00000, 000000, 000000, 00000, 000000, 000000, 00001,
000067, 000067, 00000, 000000, 000000, 000000, 003600,
01, Fri 01:00 - 02:00, Agent, 2002, 00000, 000000, 000000, 00000, 000000, 000000, 00001,
000006, 000006, 00000, 000000, 000000, 000000, 003600,
01, Fri 01:00 - 02:00, Agent, 2005, 00000, 000000, 000000, 00000, 000000, 000000, 00001,
000004, 000004, 00000, 000000, 000000, 000000, 003600,
01, Fri 01:00 - 02:00, Queue, 00007, 00005, 00000, 00013, 00030, 00002, 00034, 00002, 00000,
 00000, 00000, 00002, 00000, 00000, 00003, 00002, 00000, 00000, 00000, 00000,
01, Fri 02:00 - 03:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Fri 03:00 - 04:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
.
.
.
02, Thu 20:00 - 21:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
02, Thu 21:00 - 22:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
02, Thu 22:00 - 23:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
02, Thu 23:00 - 00:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
02, Fri 00:00 - 01:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
```

Related Commands

Command	Description
hunt-group report delay hours	Delays hunt group statistics collection for a specified number of hours.
hunt-group report every hours	Sets the hourly interval at which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco Unified CME B-ACD call statistics are to be sent using TFTP.
hunt-group statistics write-all	Writes all ephone and voice hunt group statistics to a fille.
show voicehunt-groupstatistics	Displays voice hunt group statistics.
show ephone-hunt statistics	Displays ephone hunt group statistics.

hunt-group logout

To set the hunt-group logout options, use the **hunt-group logout** command with **DND**, **Hlog**, **notify**, and **threshold** keywords in telephony-service configuration mode. To return to the default, use the **no** form of this command.

hunt-group logout [{**DND** | **HLog** | **notify** | **threshold** *number*}]
no hunt-group logout [{**DND** | **HLog** | **notify** | **threshold** *number*}]

Syntax Description

DND	Agent phones do not answer the number of calls specified in the **auto logout** command and are automatically placed in both DND status and not-ready status. The HLog soft key is not displayed on phones.
HLog	Agent phones do not answer the number of calls specified in the **auto logout** command and are automatically placed only in not-ready status. The HLog soft key is displayed on phones in addition to the DND soft key.
notify	Enables logout call in queue notification on HLog PLK button.
threshold *number*	Defines the boundary value by which how the Hlog PLK indicates the number of calls in queue on the logout agent's phone. Range is 1 to 65535.

Command Default

DND and HLog functionality is not separate and the HLog soft key will not be displayed on phones. The default is DND.

The default for threshold is disabled and the LED on the HLOG PLK blinks slow in amber as long as there are calls in queue.

The default for notify is disabled and has no LED display.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
15.2(2)T2	Cisco Unified CME 9.1	The **notify** and **threshold** keywords were added.

Usage Guidelines

When Do Not Disturb (DND) functionality is activated, no calls are received at the phone, including ephone hunt group calls. DND is activated and canceled using the DND soft key or the DND feature access code (FAC).

When HLog functionality is activated, hunt-group agents are placed in not-ready status and hunt-group calls are blocked from the phone. Other calls that directly dial the phone's extension numbers are still received at the phone. HLog is activated and canceled using the HLog soft key or an HLog FAC.

If the **auto logout** command is used, the Automatic Agent Status Not-Ready feature is invoked for an ephone hunt group. This feature is triggered when an ephone-dn member does not answer a specified number of ephone hunt group calls. The following actions take place:

- If the **hunt-group logout HLog** command has been used, the agent is placed in not-ready status. The agent's ephone-dn will not receive further hunt group calls but will receive calls that directly dial the ephone-dn's extension numbers. An agent in not-ready status can return to ready status by pressing the HLog soft key or by using the HLog FAC.
- If the **hunt-group logout HLog** command has not been used or if the **hunt-group logout DND** command has been used, the phone on which the ephone-dn appears is placed into DND mode, in which the ephone-dn does not receive any calls at all, including hunt-group calls. The red lamp on the phone lights to indicate DND status. An agent in DND mode can return to ready status by pressing the DND soft key or by using the DND FAC.

The **DND** and **Hlog** keywords are exclusive and are used to enable separate handling of DND and HLog functionality for hunt-group agents and to display the HLog softkey on phones.

The **notify** and **threshold** keywords are used to enable the indication of the calls in queue for logout agents using the Hlog Programmable Line Key.

If the **threshold** *number* is enabled, the LED on the Hlog PLK blinks slow in amber for the number of calls in queue less than the threshold and blinks fast in red for those equal or beyond the threshold. This command will not take effect if **hunt-group logout** *notify* is disabled.

Note When an agent who is a dynamic member of a hunt group is in not-ready status, the agent's slot in the ephone hunt group is not relinquished. It remains reserved by the agent until the agent leaves the group.

Examples

The following example creates hunt group 3 with three agents (extensions 1001, 1002, and 1003). It specifies that after one unanswered call, an agent should be put into not-ready status but not into DND status.

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group logout HLog
Router(config-telephony)# exit

Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# pilot 4200
Router(config-ephone-hunt)# list 1001, 1002, 1003
Router(config-ephone-hunt)# timeout 10
Router(config-ephone-hunt)# auto logout
Router(config-ephone-hunt)# final 4500
```

The following example sets the value of threshold to 2:

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group logout ?
  DND        logout using DND softkey or PLK
  HLog       logout using HLog softkey or PLK
  notify     enable logout call in queue notification on HLog PLK button
  threshold  configure logout call in queue threshold
Router(config-telephony)# hunt-group logout threshold ?
  <1-65535>  number of calls in queue
Router(config-telephony)# no hunt-group logout notify
Router(config-telephony)# no hunt-group logout threshold
% Incomplete command.

Router(config-telephony)# no hunt-group logout threshold 2
```

```
Router(config-telephony)# no hunt-group logout ?
  DND        logout using DND softkey or PLK
```

HLog logout using HLog softkey or PLK

```
  notify     enable logout call in queue notification on HLog PLK button
  threshold  configure logout call in queue threshold
Router(config-telephony)# no hunt-group logout dnd
Router(config-telephony)# no hunt-group logout hlog
```

Related Commands

Command	Description
auto logout	Enables the automatic change of an agent's ephone-dn to not-ready status after a specified number of hunt-group calls are not answered.
feature-button *index* <*feature identifier* > [**label** *label*]	Enables the feature button configuration on a line key.

hunt-group report delay hours

To delay the automatic transfer of Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) call statistics to a file, use the **hunt-group report delay hours** command in telephony-service configuration mode. To remove to the delay setting, use the **no** form of this command.

hunt-group report delay *number* **hours**
no hunt-group report delay *number* **hours**

Syntax Description

number	Number of hours by which the collection of statistics can be extended for the statistics collection periods configured with the **hunt-group report every hours** command. The range is from 1 to 10.

Command Default

Hunt-group report is delayed for one hour.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command is used for Cisco CME basic automatic call distribution (B-ACD) and auto-attendant (AA) service only.

The **hunt-group report delay hours** command is used as part of a statistics reporting configuration that allows Cisco CME B-ACD call statistics to be sent automatically to files using TFTP. For detailed information, see Cisco CME B-ACD and Tcl Call-Handling Applications .

Statistics are collected and stored (**statistics collect** command and **hunt-group report url** command) in specified intervals (**hunt-group report every hours** command). The default is for the statistics to be collected one hour after the specified interval. Because calls are counted when they end, some of the longer calls may not be counted. For example, if there is a call from 1:35 p.m. to 3:30 p.m., the interval is 1 hour, and there is no delay, TFTP will write the 1 p.m. to 2 p.m. statistics at 3 p.m. However, at 3 p.m., the 1:35 p.m. call is still active, so the call will not be counted at that time as occurring in the 1 p.m. to 2 p.m. time slot. When the call finishes at 3:30 p.m., it will be counted as occurring from 1 p.m. to 2 p.m. The **show hunt-group** command will report it, but TFTP will have already sent out its report. To include the 1:35 p.m. call, you could use the **hunt-group report delay hours** command to delay TFTP statistics reporting for an extra hour so the 1 p.m. to 2 p.m. report will be written at 4 p.m. instead of 3 p.m.

Examples

The following example shows a configuration in which statistics are reported for B-ACD calls that occur within three-hour time frames, but the collection of the statistic collection is extended for an extra hour to include calls that did not end within the three-hour time period:

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group report every 3 hours
Router(config-telephony)# hunt-group report delay 1 hours
```

The following is an example of a report that the previous configuration might send to a file if the **statistics collect** command was entered at 18:20:

```
23:00:00 UTC Tue Dec 20 2004,
,
01, Tue 18:00 - 19:00, HuntGp, 02, 01, 00005, 00002, 0003, 0006, 000001, 000001, 0011,
01, Tue 19:00 - 20:00, HuntGp, 02, 02, 00000, 00000, 0000, 0000, 000000, 000000, 0000,
01, Tue 20:00 - 21:00, HuntGp, 02, 02, 00006, 00003, 0003, 0009, 000001, 000003, 0012,
```

Statistics collection has to take place for at least three hours for the statistics to be written to a file. The following is a chronology of events:

- At 19:00, the statistics collection was active for 40 minutes, so no statistics were written to file.
- At 20:00, the statistics collection was active for 1 hour and 40 minutes, so no statistics were written to file.
- At 21:00, the statistics collection was active for 2 hours and 40 minutes, so no statistics were written to file.
- At 22:00, the statistics collection was active for 3 hours and 40 minutes but there is a one-hour delay, so no statistics were written to file.
- At 23:00 the statistics were written to a file using TFTP.

Related Commands

Command	Description
hunt-group report every hours	Sets the hourly interval after which Cisco CME B-ACD call statistics are automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco CME B-ACD call statistics are to be sent using TFTP.
statistics collect	Enables the collection of Cisco CME B-ACD call data for an ephone hunt group.

hunt-group report every hours

To set the hourly interval at which Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) call statistics are automatically transferred to a file, use the **hunt-group report every hours** command in telephony-service configuration mode. To remove the interval setting, use the **no** form of this command.

hunt-group report every *number* **hours**
no hunt-group report every *number* **hours**

Syntax Description

number	Number of hours after which auto-attendant (AA) call statistics are collected and reported. The range is from 1 to 84.

Command Default

No hourly interval is configured.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command is used for Cisco CME basic automatic call distribution (B-ACD) and auto-attendant (AA) service only.

The **hunt-group report every hours** command is used as part of a statistics reporting configuration that allows Cisco CME B-ACD call statistics to be sent automatically to files by means of TFTP. For detailed information, see Cisco CME B-ACD and Tcl Call-Handling Applications .

Because calls are counted when they end, some of the longer calls may not be counted in the report. To delay the time in which statistics are collected and transferred you may configure a delay time with the **hunt-group report delay hours** command.

Examples

The following example sets the statistics collection to occur every three hours. There is no delay.

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group report every 3 hours
```

The following is an example of a report that the previous configuration might send to a file if the **statistics collect** command was entered at 18:20:

```
22:00:00 UTC Tue Dec 20 2005,
,
01, Tue 18:00 - 19:00, HuntGp, 02, 01, 00005, 00002, 0003, 0006, 000001, 000001, 0011,
01, Tue 19:00 - 20:00, HuntGp, 02, 02, 00000, 00000, 0000, 0000, 000000, 000000, 0000,
01, Tue 20:00 - 21:00, HuntGp, 02, 02, 00006, 00003, 0003, 0009, 000001, 000003, 0012,
```

Statistics collection has to take place for at least three hours for the statistics to be written to a file. The following is a chronology of events:

- At 19:00, the statistics collection was active for 40 minutes, so no statistics were written to file.

- At 20:00, the statistics collection was active for 1 hour and 40 minutes, so no statistics were written to file.
- At 21:00, the statistics collection was active for 2 hours and 40 minutes, so no statistics were written to file.
- At 22:00, the statistics collection was active for 3 hours and 40 minutes, so statistics were written to a file using TFTP.

If the previous example were configured for a delay of one hour using the **hunt-group report delay 1 hours** command, the statistics would be written one hour later at 23:00.

Related Commands

Command	Description
hunt-group report delay hours	Delays the automatic transfer of Cisco CME B-ACD call statistics to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco CME B-ACD call statistics are to be sent using TFTP.
statistics collect	Enables the collection of Cisco CME B-ACD call statistics for an ephone hunt group.

hunt-group statistics write-all

To write all the ephone and voice hunt group statistics to a file, use the **hunt-group statistics write-all** command in privileged EXEC mode.

hunt-group statistics write-all *location*

Syntax Description

location	URL or filename to which the statistics should be written.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
15.2(2)T	Cisco Unified CME 9.0	This command was introduced to replace the **ephone-hunt statistics write-all** command.

Usage Guidelines

Use the **hunt-group statistics write-all** command to write out, in hourly increments, all the ephone and voice hunt group statistics for the past seven days. This command is intended to be used when normal hunt group statistics collection is interrupted, perhaps due to TFTP server failure. If applicable, the TFTP statistics report consists of both ephone and voice hunt statistics.

The commands that are normally used to provide hunt group statistics are **hunt-group report delay hours**, **hunt-group report every hours**, **hunt-group report url**, and **statistics collect**. These commands allow you to specify shorter, more precise reporting periods and file naming conventions.

Note: On the day that daylight saving time adjusts the time back by one hour at 2 a.m. each year, the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

Examples

The following example shows how the **hunt-group statistics write-all** command writes a combination of ephone and voice hunt group statistics to a file in TFTP server 223.255.254.254:

```
Router# hunt-group statistics write-all tftp://223.255.254.254/ngm/huntgp/uc500/statall
Writing out all hunt group statistics to tftp://223.255.254.254/ngm/huntgp/uc500/statall
00:08:34 UTC Sat Feb 19 2011,
,
EPHONE HUNT GROUP STAT,
01, Sat 00:00 - 01:00, HuntGp, 02, 02, 00001, 00001, 00000, 0007, 0007, 000001, 000001,
0000, 00000, 000000, 000000,
01, Sat 00:00 - 01:00, Agent, 5012, 00001, 000001, 000001, 00000, 000000, 000000, 00001,
000008, 000008, 00001, 000003, 000003,
01, Sat 00:00 - 01:00, Queue, 00000, 00001, 00000, 00006, 00006, 00000, 00000, 00000, 00000,
 00000,
01, Sat 01:00 - 02:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 02:00 - 03:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
```

```
0000, 00000, 000000, 000000,
01, Sat 03:00 - 04:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 04:00 - 05:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 05:00 - 06:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 06:00 - 07:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 07:00 - 08:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000,
.
.
.
06, Fri 19:00 - 20:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
06, Fri 20:00 - 21:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
06, Fri 21:00 - 22:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
06, Fri 22:00 - 23:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
06, Fri 23:00 - 00:00, HuntGp, 02, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
VOICE HUNT GROUP STAT,
01, Sat 00:00 - 01:00, HuntGp, 08, 08, 00002, 00002, 00000, 0002, 0003, 000004, 000005,
0000, 00001, 000003, 000003,
01, Sat 00:00 - 01:00, Agent, 5022, 00001, 000005, 000005, 00000, 000000, 000000, 00000,
000000, 000000, 00000, 000000, 000000,
01, Sat 00:00 - 01:00, Agent, 5012, 00001, 000004, 000004, 00001, 000003, 000003, 00001,
000005, 000005, 00001, 000003, 000003,
01, Sat 00:00 - 01:00, Queue, 00001, 00001, 00000, 00003, 00003, 00000, 00000, 00000, 00000,
 00000,
01, Sat 01:00 - 02:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 02:00 - 03:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 03:00 - 04:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 04:00 - 05:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Sat 05:00 - 06:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
.
.
.
08, Fri 19:00 - 20:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
08, Fri 20:00 - 21:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
08, Fri 21:00 - 22:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
08, Fri 22:00 - 23:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
08, Fri 23:00 - 00:00, HuntGp, 08, 08, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
```

Related Commands

Command	Description
hunt-group report delay hours	Delays hunt group statistics collection for a specified number of hours.

Command	Description
hunt-group report every hours	Sets the hourly interval at which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco Unified CME B-ACD call statistics are to be sent using TFTP.
show ephone-hunt	Displays ephone hunt group information.
show ephone-hunt statistics	Displays ephone hunt group statistics.
statistics collection	Enables the collection of call statistics for an ephone hunt group.
statistics collection (voice hunt-group)	Enables the collection of call statistics for a voice hunt group.

huntstop (ephone-dn and ephone-dn-template)

To disable call hunting for directory numbers or channels, use the **huntstop** command in ephone-dn or ephone-dn-template configuration mode. To reset to the default, use the **no** form of this command.

huntstop [**channel** *number-of-channels*]
no huntstop [**channel** *number-of-channels*]

Syntax Description

channel	(Optional) For dual-line and octo-line directory numbers. Prevents incoming calls from hunting to the next channel if the first channel is busy or does not answer.
number-of-channels	Supported for octo-line directory numbers only. Number of channels available to accept incoming calls. Remaining channels are reserved for outgoing calls or features such as call transfer, call waiting, and conferencing. Range: 1 to 8. Default: 8.

Command Default

Ephone-dn huntstop is enabled. Channel huntstop is disabled for dual-line directory numbers. Channel huntstop is set to 8 for octo-line directory numbers.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)ZJ	Cisco CME 3.0	The **channel** keyword was introduced.
12.3(4)T	Cisco CME 3.0	This **channel** keyword was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was added to ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.
12.4(15)XZ	Cisco Unified CME 4.3	The *number-of-channels* argument was added for octo-lines.
12.4(20)T	Cisco Unified CME 7.0	This command with the *number-of-channels* argument for octo-lines was integrated into Release 12.4(20)T.

Usage Guidelines

Use this command without the **channel** keyword to disable call hunting for ephone-dns. An incoming call does not roll over (hunt) to another ephone-dn if the called number is busy or does not answer and a call hunt strategy has been established that includes this ephone-dn. A huntstop prevents hunt-on-busy from redirecting a call from a busy phone into a dial-peer with a catch-all default destination. Use the **no huntstop** command to disable huntstop and allow hunting for ephone-dns.

Channel huntstop works in a similar way, but it affects call hunting behavior for the two channels of a dual-line ephone-dn. Use the **channel** keyword to prevent incoming calls from hunting to the second channel of an ephone-dn if the first channel is busy or does not answer. Incoming calls hunt forward to the next ephone-dn in the hunt sequence instead of to the next channel on the same ephone-dn.

For example, an incoming call might search through the following ephone-dns and channels:

ephone-dn 10 (channel 1) ephone-dn 10 (channel 2)

ephone-dn 11 (channel 1) ephone-dn 11 (channel 2) ephone-dn 12 (channel 1) ephone-dn 12 (channel 2)

If the **huntstop channel** command is not enabled (the default), a call might ring for 30 seconds on ephone-dn 10 (channel 1) and then after 30 seconds move to ephone-dn 10 (channel 2), which is usually not the desired behavior. It is useful to reserve the second channel of a dual-line ephone-dn for call transfer, call waiting, or conferencing.

The *number* argument is required for an octo-line directory number when using the **channel** keyword. This argument limits the number of channels for incoming calls on an octo-line directory number and reserves the other channels for outgoing calls or features such as call transfer or conferencing. The router selects idle channels from the lowest number to the highest. This argument is supported only for an octo-line directory number.

In an ephone-dn template, you can apply separate **huntstop channel** commands for dual-line directory numbers and octo-line directory numbers.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example shows huntstop is disabled for ephone-dn 1. The huntstop attribute is set to OFF and allows calls to extension 5001 to hunt to another directory number when directory number 1 is busy.

```
ephone-dn 1
 number 5001
 no huntstop
```

The following example shows a typical configuration in which enabling huntstop (default) is required:

```
ephone-dn 1
 number 5001

ephone 4
 button 1:1
 mac-address 0030.94c3.8724

dial-peer voice 5000 voip
 destination-pattern 5...
 session target ipv4:192.168.17.225
```

In the previous example, the huntstop attribute for the dial peer is set to ON by default and prevents calls to extension 5001 from being rerouted to the on-net H.323 dial peer for 5... when extension 5001 is busy (the three periods are used as wildcards).

The following example shows another configuration in which huntstop is not desired and is explicitly disabled. In this example, ephone 4 is configured with two lines, each with the same extension number 5001. This allows the second line to provide call-waiting notification for extension number 5001 when the first line is in use. Setting **no huntstop** on the first line (ephone-dn 1) allows incoming calls to hunt to the second line (ephone-dn 2) when the first line is busy.

Ephone-dn 2 has call forwarding set to extension 6000, which corresponds to a locally attached answering machine connected to a foreign exchange station (FXS) voice port. In this example, the plain old telephone system (POTS) dial peer for extension 6000 also has the dial-peer huntstop attribute explicitly set to prevent further hunting.

```
ephone-dn 1
 number 5001
 no huntstop
 preference 1
 call-forward noan 6000

ephone-dn 2
 number 5001
 preference 2
 call-forward busy 6000
 call-forward noan 6000

ephone 4
 button 1:1 2:2
 mac-address 0030.94c3.8724

dial-peer voice 6000 pots
 destination-pattern 6000
 huntstop
 port 1/0/0
 description answering-machine
```

The following example shows a dual-line configuration in which an ephone-dn template is used to prevent calls from hunting to the second channel of any ephone-dn. The calls hunt through the first channels for each ephone-dn in the order 10, 11, 12.

```
ephone-dn-template 2
 huntstop channel

ephone-dn 10 dual-line
 number 1001
 no huntstop
 ephone-dn-template 2

ephone-dn 11 dual-line
 number 1001
 no huntstop
 ephone-dn-template 2
 preference 1

ephone-dn 12 dual-line
 number 1001
 no huntstop
 ephone-dn-template 2
 preference 2
```

The following example shows a configuration in which incoming calls to octo-line directory number 7 are limited to four, freeing the other four channels for outgoing calls or features such as call transfer or conferencing.

```
ephone-dn  7  octo-line
 number 2001
 name Smith, John
 huntstop channel 4
```

The following example shows an ephone-dn template configuration in which the huntstop is set for both dual-line and octo-line directory numbers.

```
ephone-dn-template  1
 huntstop channel
 huntstop channel 4
```

Related Commands

Command	Description
huntstop (dial-peer)	Disables further dial-peer hunting if a call fails using hunt groups.
number	Associates a telephone or extension number with a directory number (ephone-dn).

huntstop (voice register dn)

To disable call hunting behavior for a directory number on a SIP phone, use the **huntstop** command in voice register dn configuration mode. To reset to the default, use the **no** form of this command.

huntstop [**channel** *number*]
no huntstop [**channel** *number*]

Syntax Description

channel *number*	(Optional) Number of channels available to accept incoming calls. Remaining channels are reserved for outgoing calls or features such as call transfer, call waiting, and conferencing. Range: 1 to 50. Default: 0 (disabled).

Command Default

Call hunting is enabled for the directory number. Channel huntstop is disabled (0) for the directory number.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(22)YB	Cisco Unified CME 7.1	The **channel** keyword and *number* argument were added.
12.4(24)T	Cisco Unified CME 7.1	This command has been integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command disables call hunting behavior for a directory number on a SIP IP phone so that an incoming call does not roll over (hunt) to another directory number if the called directory number is busy or does not answer and if a hunting strategy has been established that includes this directory number. A huntstop allows you to prevent hunt-on-busy from redirecting a call from a busy phone into a dial-peer setup with a catch-all default destination. Use the **no huntstop** command to disable huntstop and allow hunting for directory numbers (default).

The **channel** keyword and *number* argument limits the number of channels for incoming calls to a directory number and reserves the other channels for outgoing calls or features such as call transfer or conferencing. The router selects idle channels from the lowest number to the highest.

Examples

The following example shows a typical configuration in which huntstop is required. The **huntstop** command is enabled and prevents calls to extension 5001 from being rerouted to the on-net H.323 dial peer for 5... when extension 5001 is busy (three periods are used as wild cards).

```
voice register dn 1
 number 5001
 huntstop

voice register pool 4
 button 1:1
 mac-address 0030.94c3.8724

dial-peer voice 5000 voip
```

```
 destination-pattern 5...
 session target ipv4:192.168.17.225
```

The following example shows a configuration in which huntstop is not desired (default). In this example, directory number 4 is configured with two lines, each with the same extension number 5001. This is done to allow the second line to provide call-waiting notification for extension number 5001 when the first line is in use. Not enabling huntstop on the first line (directory number 1) allows incoming calls to hunt to the second line (directory number 2) on phone 4 when the directory number 1 line is busy.

Directory number 2 has call forwarding set to extension 6000, which corresponds to a locally attached answering machine connected to a foreign exchange station (FXS) voice port. In this example, the plain old telephone system (POTS) dial peer for extension 6000 has the dial-peer huntstop attribute explicitly set to prevent further hunting.

```
voice register dn 1
 number 5001
 preference 1
 call-forward noan 6000

voice register dn 2
 number 5001
 preference 2
 call-forward busy 6000
 call-forward noan 6000

voice register pool 4
 button 1:1 2:2
 mac-address 0030.94c3.8724

dial-peer voice 6000 pots
 destination-pattern 6000
 huntstop
 port 1/0/0
 description answering-machine
```

The following example shows a configuration in which incoming calls to directory number 23 are blocked if the total number of calls to extension 8123 exceeds 4. This frees the other channels for outgoing calls or features such as call transfer or conferencing.

```
voice register dn  23
 number 8123
 shared-line max-calls 4
 huntstop channel 4
```

Related Commands

Command	Description
huntstop (dial-peer)	Disables all further dial-peer hunting if a call fails on the dial peer.
shared-line	Creates a directory number to be shared by multiple SIP phones.

Cisco Unified CME Commands: I

ica

To specify the audio file used for the isolated code announcement, use the **ica** command in voice MLPP configuration mode. To disable use of this audio file, use the **no** form of this command.

ica *audio-url*
no ica

Syntax Description

audio-url	Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.

Command Default

No announcement is played.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that plays to callers when service or equipment problems prevent completion of their call.

The **mlpp indication** command must be enabled (default) for a phone to play precedence announcements.

This command is not supported by Cisco IOS help. If you type **?** , Cisco IOS help does not display a list of valid entries.

Examples

The following example shows that the audio file played for the isolated code announcement is named ica.au located in flash:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# ica flash:ica.au
```

Related Commands

Command	Description
bnea	Specifies the audio file used for the busy station not equipped for preemption announcement.
upa	Specifies the audio file used for the unauthorized precedence announcement.
vca	Specifies the audio file used for the vacant code announcement.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.

id (voice register pool)

To explicitly identify a locally available individual Cisco SIP IP phone, or when running Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST), set of Cisco SIP IP phones, use the **id** command in voice register pool configuration mode. To remove local identification, use the **no** form of this command.

id {**[network** *address* **mask** *mask* **|** *address* **mask** *mask* **] | [ip** *address* **mask** *mask* *address* **mask** *mask* **] | [mac** *address*]}**[device-id-name** *devicename*]
no id {**[network** *address* **mask** *mask* **|** *address* **mask** *mask* **] | [ip** *address* **mask** *mask* *address* **mask** *mask* **] | [mac** *address*]}**[device-id-name** *devicename*]

Syntax Description

network *address* **mask** *mask* | *address* **mask** *mask*	This keyword/argument combination is used to accept SIP Register messages for the indicated phone numbers from any IP phone within the specified IPv4 and IPv6 subnets. *ipv6 address* can only be configured with an IPv6 address or a dual-stack mode.
ip *address* **mask** *mask* | *address* **mask** *mask*	This keyword/argument combination is used to identify an individual phones IPv4 or IPv6 address. *ipv6 address* can only be configured with an IPv6 address or a dual-stack mode.
mac *address*	The **mac** *address* keyword/argument combination is used to identify the MAC address of a particular Cisco IP phone.
device-id-name *devicename*	Defines the device name to be used to download the phone’s configuration file.

Command Default

No SIP IP phone is configured.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.
15.3(3)T	Cisco Unified CME 10.0	This command was modified to add the **device-id-name** *devicename* keyword-argument combination.
Cisco IOS XE Everest 16.6.1	Unified SRST 12.0	This command was modified to add the following keyword-argument combinations for **network** and **ip** to include support for IPv6 address: *address* **mask** *mask*.

Usage Guidelines

Configure this command before configuring any other command in voice register pool configuration mode.

This command allows explicit identification of an individual Cisco SIP IP phone to support a degree of authentication, which is required to accept registrations, based upon the following:

- Verification of the local Layer 2 MAC address using the router's Address Resolution Protocol (ARP) cache.
- Verification of the known single static IP address (or DHCP dynamic IP address within a specific subnet) of the Cisco SIP IP phone.

When the **mac** *address* keyword and argument are used, the IP phone must be in the same subnet as that of the router's LAN interface, such that the phone's MAC address is visible in the router's ARP cache. Once a MAC address is configured for a specific voice register pool, remove the existing MAC address before changing to a new MAC address.

Note For Cisco Unified SIP SRST, this command also allows explicit identification of locally available set of Cisco SIP IP phones.

Examples

The following is partial sample output from the **show running-config** command. The **id** command identifies the MAC address of a particular Cisco IP phone. The output shows that voice register pool 1 has been set up to accept SIP Register messages from a specific IP phone through the use of the **id** command.

```
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
 proxy 10.2.161.187 preference 1 monitor probe icmp-ping
 alias 1 94... to 91011 preference 8
 voice-class codec 1
```

The following is sample output from the **show running-config** command after configuring IPv6 address on Cisco Unified SRST router.

```
voice register pool 1
 id network 2001:420:54FF:13::312:0/117
```

Related Commands

Command	Description
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CallManager Express (Cisco Unified CME) system.

import certificate

To import a trusted certificate in PEM format from flash memory to the CTL file of an IP phone, use the **import certificate** command in ctl-client configuration mode. To return to the default, use the **no** form of this command.

import certificate *tag description***flash:***cert_name*
no import certificate

Syntax Description

tag	Identifier for the trusted certificate.
description	Descriptive name of the trusted certificate.
flash:*cert_name*	Specifies the filename of the trusted certificate stored in flash memory.

Command Default

None

Command Modes

CTL-client configuration (config-ctl-client)

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

A CTLFile.tlv file should appear in the flash location after using the **regenerate** command in ctl-client configuration mode. If the file is missing, use the **debug ctl-client** command, followed by the **regenerate** command.

Examples

The following is an example of how the **import certificate** command is used to import the WebServer certificate with filename web_cer.cer from flash memory:

```
Router(config)# ctl-client
Router(config-ctl-client)# sast1 trustpoint primary-cme
Router(config-ctl-client)# sast2 trustpoint sast-secondary
Router(config-ctl-client)# import certificate 1 WebServer flash:web_cert.cer
Router(config-ctl-client)# regenerate
```

Related Commands

Command	Description
ctl-client	Enters CTL-client configuration mode to set parameters for the CTL client.

index (lpcor ip-phone)

To add a logical partitioning class of restriction (LPCOR) group to the IP-phone subnet table, use the **index** command in LPCOR ip-phone subnet configuration mode. To remove a resource, use the **no** form of this command.

index *index-number lpcor-group* {*ipv4-address network-mask* [**vrf** *vrf-name*] | **dhcp-pool** *pool-name*}
no index *index-number*

Syntax Description

index-number	Number of the LPCOR subnet index entry. Range: 1 to 50.
lpcor-group	Name of a LPCOR resource-group policy.
ipv4-address	IPv4 address of the LPCOR policy.
network-mask	Subnet mask for the associated IPv4 address.
vrf *vrf-name*	(Optional) Dynamic Host Configuration Protocol (DHCP) server uses the VPN routing and forwarding (VRF) table that is associated with the access point name (APN).
dhcp-pool *pool-name*	User-defined name of the DHCP pool. The pool name can be a symbolic string (such as Sales) or an integer (such as 0).

Command Default

No index entry is configured.

Command Modes

LPCOR ip-phone subnet configuration (cfg-lpcor-ipphone-subnet)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command is used for mobility-type phones only, which can include Extension Mobility phones, teleworker remote phones, and Cisco IP Communicator softphones.

Two IP-phone subnet tables, containing up to 50 index entries, can be defined on each Cisco Unified CME router. One table is for incoming calls and the other table is for outgoing calls.

A LPCOR policy is dynamically associated with calls to and from a mobility-type phone by matching its current IP address or DHCP pool in the IP-phone subnet table. If the LPCOR policy cannot be provisioned from the IP-phone subnet table, the default LPCOR policy for mobility-type phones is used.

Entries in the IP-phone subnet tables are indexed in ascending order. The lookup of entries is in sequential ascending order. After Cisco Unified CME finds a matching entry, the corresponding LPCOR policy is associated with the call. Even if there are other entries that are a better match, only the first match is used.

For instance, in the example below, if a call originates from an IP phone with IP address 10.1.10.3, LPCOR policy local_g4 is associated with the incoming call instead of LPCOR policy local_g5 even though local_g5 is a better match.

Examples

The following example shows an IP-phone subnet table for incoming calls that has four entries:

```
voice lpcor ip-phone subnet incoming
 index 1 local_g4 10.1.10.0 255.255.255.0
 index 2 remote_g4 171.19.0.0 255.255.0.0
 index 3 local_g5 10.1.10.2 255.255.255.255
 index 4 local_g5 10.1.10.3 255.255.255.255
```

Related Commands

Command	Description
lpcor type	Specifies the LPCOR type for an IP phone.
voice lpcor ip-phone mobility	Sets the default LPCOR policy for mobility-type phones.
voice lpcor policy	Creates a LPCOR policy for a resource group.

index (lpcor ip-trunk)

To add a logical partitioning class of restriction (LPCOR) resource group to the IP trunk subnet table, use the **index** command in LPCOR IP-trunk subnet configuration mode. To remove a resource, use the **no** form of this command.

index *number* *lpcor-group* {*ipv4-address* *network-mask* | **hostname** *host-name*}
no **index** *number*

Syntax Description

number	Number of the LPCOR subnet index entry. Range: 1 to 50.
lpcor-group	Name of a LPCOR resource-group policy.
ipv4-address	IPv4 address of the LPCOR policy.
network-mask	Subnet mask of the associated IPv4 address.
hostname *host-name*	User-defined IP host name.

Command Default

No index entry is configured.

Command Modes

LPCOR IP-trunk subnet configuration (cfg-lpcor-iptrunk-subnet)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

One IP-trunk subnet table, containing up to 50 index entries, can be defined on each Cisco Unified CME router for incoming VoIP trunk calls (H.323 or SIP).

An incoming VoIP trunk call is associated with a LPCOR policy by matching the remote IP address to an entry in the incoming IP-trunk subnet table. If that is not successful, the LPCOR policy in voice service configuration mode is applied.

Entries in the IP-trunk subnet table are indexed in ascending order. The lookup of entries is in sequential ascending order. After Cisco Unified CME finds a matching entry, it associates the corresponding LPCOR policy with the call. Even if there are other entries that are a better match, only the first match is used.

In the following example, an incoming VoIP call with a remote IP address of 172.19.22.25 is associated with sip_group1 even though voip_group2 is a better match.

Examples

The following example shows an IP-trunk subnet table with six index entries:

```
voice lpcor ip-trunk subnet incoming
 index 1 h323_group1 172.19.33.0 255.255.255.0
 index 2 sip_group1 172.19.22.0 255.255.255.0
 index 3 voip_group2 172.19.33.25 255.255.255.255
 index 4 voip_group3 172.19.22.26 255.255.255.255
```

```
index 5 sip_s1 hostname sipserver1
index 6 sip_s2 hostname sipserver2
```

Related Commands

Command	Description
lpcor incoming	Associates an incoming call with a LPCOR resource-group policy.
voice lpcor policy	Ccreates a LPCOR policy for a resource group.

intercom (ephone-dn)

To create an intercom by programming a pair of extensions (ephone-dns) to automatically call and answer each other, use the **intercom** command in ephone-dn configuration mode. To remove an intercom, use the **no** form of this command.

intercom *extension-number* **[[barge-in [no-mute] |no-auto-answer | no-mute] [label***label***]] |**
label*label***]|[***paging number***ptt]**
no intercom

Syntax Description

extension-number	Extension or telephone number to which calls are placed.
barge-in	(Optional) Allows inbound intercom calls to force an existing call into the call-hold state and the intercom call to be answered immediately.
label *label*	(Optional) Defines an alphanumeric label for the intercom, of up to 30 characters.
no-auto-answer	(Optional) Disables the intercom auto-answer feature.
no-mute	(Optional) Allows an intercom call to be answered without deactivating a speaker's mute key.
paging number ptt	(Optional) Allows to set a paging number for push-to-talk (PTT) feature.

Command Default

Intercom functionality is disabled.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.3(11)XL	Cisco CME 3.2.1	The **no-mute** keyword was added.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(20)T	Cisco Unified CME 7.0	The paging number and ptt keywords and argument was added.

Usage Guidelines

This command is used to dedicate a pair of Cisco ephone-dns for use as a "press to talk" two-way intercom between Cisco IP phones. Intercom lines cannot be used in shared-line configurations. If an ephone-dn is configured for intercom operation, it must be associated with one Cisco IP phone only. The intercom attribute causes an IP extension (ephone-dn) to operate in autodial fashion for outbound calls and autoanswer-with-mute for inbound calls.

The **barge-in** keyword allows inbound intercom calls to force an existing call on the called phone into the call-hold state to allow the intercom call to be answered immediately. The **no-auto-answer** keyword creates for the IP phone line a connection that resembles a private line, automatic ringdown (PLAR). The **label** keyword defines a text label for the intercom.

Following this command, the intercom ephone-dns are assigned to ephones using the **button** command. Following the **button** command, the **restart** command must be used to initiate a quick reboot of the phones to which this intercom is assigned.

The default **intercom** command behavior is speakers are set to mute automatically when phones receive intercom calls. For example, if phone user 1 places an intercom call and connects to phone user 2, user 2 will hear user 1, but user 1 will not hear user 2. To be heard, user 2 must first disable the speaker's mute function. The benefit is people who receive intercom calls can use the mute button to control when they will be heard initially.

The **no-mute** keyword deactivates the speaker mute function when IP phones receive intercom calls. For example, if phone user 1 makes an intercom call to phone user 2, both users will hear each other upon connection. The benefit is that people who receive intercom calls do not have to disable their speaker's mute function to be heard, *but* their conversations and nearby background sounds will be heard the moment an intercom call to them is connected—regardless of whether they are ready to take a call or not.

The intercom command allows you to add a paging number to behave as a push-to-talk (ptt) feature. More information on the push-to-talk feature is available at this link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmelabel.html#wpmkr1048855

Examples

The following example sets the intercom on Cisco IP phone directory number 1:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number A5001
Router(config-ephone-dn) name "intercom"
Router(config-ephone-dn) intercom A5002 barge-in
```

The following example shows intercom configuration between two Cisco IP phones:

```
ephone-dn 18
 number A5001
 name "intercom"
 intercom A5002 barge-in
ephone-dn 19
 number A5002
 name "intercom"
 intercom A5001 barge-in
ephone 4
 button 1:2 2:4 3:18
 ephone 5
 button 1:3 2:6 3:19
```

In the example, ephone-dn 18 and ephone-dn 19 are set as an intercom pair. Ephone-dn 18 is associated with button 3 of Cisco IP phone (ephone) 4, and ephone-dn 19 is associated with button number 3 of Cisco IP phone (ephone) 5. Button 3 on Cisco IP phone 4 and button 3 on Cisco IP phone 5 are set as a pair to provide intercom service to each other.

The intercom feature acts as a combination speed-dial PLAR and autoanswer with mute. If the **barge-in** keyword is set on the ephone-dn that receives the intercom call, the existing call is forced into the hold state, and the intercom call is accepted. If the phone user has the handset off hook (that is, not in speakerphone mode), the user hears a warning beep, and the intercom call is immediately connected with two-way audio. If the phone user is using speakerphone mode, the intercom connects with the microphone mute activated.

Note Any caller can dial in to an intercom extension, and a call to an intercom extension that is originated by a nonintercom caller triggers an automatic answer exactly like a legitimate intercom call. To prevent nonintercom originators from manually dialing an intercom destination, you can use alphabetic characters when you assign numbers to intercom extensions using the **number** command. These characters cannot be dialed from a normal phone but can be dialed by preprogrammed intercom extensions whose calls are made by the router.

Related Commands

Command	Description
button	Associates ephone-dns with individual buttons on Cisco IP phones and specifies ring behavior per button.
number	Associates a telephone or extension number with an extension (ephone-dn).
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

intercom (voice register dn)

To enable the intercom call option on a Cisco Unified SIP IP phone, use the **intercom** command in voice register dn configuration mode. To prevent a Cisco Unified SIP IP phone from making an intercom call, use the **no** form of this command.

intercom [**speed-dial** *digit-string*] [**label** *label-text*]
no intercom [**speed-dial** *digit-string*] [**label** *label-text*]

Syntax Description

speed-dial	(Optional) Enables the intercom line user to place a call to a pre-configured destination. If the speed-dial is not configured, it simply initiates a new call on the intercom line and waits for the user to dial the destination number.
digit-string	Digits to be dialed when the speed-dial button is pressed on a Cisco Unified SIP IP phone. For Cisco Unified SIP IP phones, if the first character of the string is a plus sign (+), the speed-dial number is locked and cannot be changed at the phone. If the only character in the string is a pound sign (#), the user-programmable speed-dial button with no speed-dial number attached is defined.
label *label-text*	(Optional) String that contains identifying text to be displayed next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.

Command Default

The Cisco Unified SIP IP phone cannot make or receive an intercom call.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

The intercom line cannot be the primary line of a Cisco Unified SIP IP phone and cannot be shared among Cisco Unified SIP IP phones.

When the intercom speed-dial option is not configured, the intercom line waits for the user to dial the destination number.

Examples

The following example shows SIP intercom configured on extension 1001:

```
Router(config)# voice register dn  1
Router(config-register-dn) number 1001
Router(config-register-dn) intercom [speed-dial 1002] [label intercom1001]
Router(config)# voice register pool 1
Router(config-register-pool) id mac 001D.452D.580C
Router(config-register-pool) type 7962
Router(config-register-pool) number 1 dn 2
Router(config-register-pool) number 2 dn 1
```

Related Commands

Command	Description
voice register dn	Enters voice register dn configuration mode.
voice register pool	Enters voice register pool configuration mode.

internal-call

To assign an MOH group for calls from an internal directory number, use the **internal-call** command in telephony-service configuration mode. To disable the internal-call command, use the **no** form of this command.

internal-call moh-group-tag
no internal-call

Syntax Description

moh-group-tag	Specifies a MOH-group number to be used for calls from an internal directory number. Range is from 0 to 5, where 0 represents MOH configuration in telephony-service configuration mode.

Command Default

No internal-call is configured.

Command Modes

Telephony-service configuration (config-telephony-service)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Before using this command make sure you have MOH-groups configured under voice-moh-group configuration mode. This command allows you to assign a MOH-group for all calls from an internal directory number. MOH group tag identifies the unique number assigned to a MOH group. Range for MOH group tag is from 0 to 5, where 0 represents MOH configuration in telephony service.

Examples

The following example shows MOH-group 4 assigned for an internal directory number:

```
telephony-service
 internal-call moh-group 4
 em logout 0:0 0:0 0:0
 max-ephones 58
 max-dn 192
 ip source-address 15.1.0.161 port 2000
 max-conferences 8 gain -6
 moh music-on-hold.au
 multicast moh 239.1.1.1 port 2000
 transfer-system full-consult
```

Related Commands

Command	Description
voice-moh-group	Enter voice-moh-group configuration mode.
moh filename	Enables music on hold from a flash audio feed
multicast moh	Enables multicast of the music-on-hold audio stream.
extension-range	Specifies the extension range for a clients calling a voice-moh-group.

ip address trusted authenticate

To enable ip address trusted authentication for incoming VoIP (H.323/SIP) calls, use the **ip address trusted authenticate** command in voice service voip mode. To disable ip address trusted authentication, use the no form of this command.

ip address trusted authenticate
no ip address trusted authenticate

Syntax Description

This command has no arguments or keywords.

Command Default

IP address trusted list authenticate is enabled.

Command Modes

Voice Service Voip

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use this command to enable the ip address trusted authentication for incoming H.323 or SIP trunk calls for toll fraud prevention on Cisco Unified CME.

Examples

The following is a sample output from this command displaying IP address trusted authentication enabled for incoming calls:

```
IP Address Trusted Authentication
 Administration State: UP
 Operation State:      UP
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 Session Targets:
Peer Tag        Oper State      Session Target
--------        ----------      --------------
11              DOWN            ipv4:1.3.45.1
1               UP              ipv4:1.3.45.1
IP Address Trusted List:
 ipv4 172.19.245.1
 ipv4 172.19.247.1
 ipv4 172.19.243.1
 ipv4 171.19.245.1
 ipv4 171.19.10.1
```

Related Commands

Command	Description
ip address trusted list	Allows to manually add additional valid IP addresses.
ip address trusted call- block cause	Allows to issues a cause-code when the incoming call is rejected by the IP address trusted authentication.

ip address trusted call-block cause

To issues a cause-code when the incoming call is rejected by the IP address trusted authentication, use the **ip address trusted call-block cause** command in voice service voip mode. To stop the IP address trusted authentication process from sending a call-block cause, use the no form of this command.

ip-address trusted call-block cause code-id
no ip-address trusted call-block cause code-id

Syntax Description

code-id	Q.850 call-disconnect cause code. Range is from 1 to 127.

Command Default

A call-reject (21) cause-code is issued to disconnect the incoming VoIP calls.

Command Modes

Voice Service voip.

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use this command to issue a cause-code when the incoming call is rejected by the IP address trusted authentication. You can issue a specific call-block cause code using any one of the Q.850 call reject cause codes.

Examples

The following is a sample output from this command displaying the default call block cause code:

```
Router #show ip address trusted list
IP Address Trusted Authentication
 Administration State: UP
 Operation State:      UP
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 Session Targets:
Peer Tag        Oper State      Session Target
--------        ----------      --------------
11              DOWN            ipv4:1.3.45.1
1               UP              ipv4:1.3.45.1
```

Related Commands

Command	Description
ip address trusted list	Allows to manually add additional valid IP addresses.
ip address trusted authenticate	Enables IP address trusted authentication for incoming VoIP calls.

ip address trusted list

To manually add multiple IP addresses for incoming VoIP (H.323/SIP) calls, use the **ip address trusted list** command in voice service voip mode. To turn off the list, use the no form of this command.

ip address trusted list ipv4 ipv4 address network mask
no ip address trusted list ipv4 ipv4 address network mask

Syntax Description

ipv4-address	IPv4 address of the incoming H.323 or SIP calls.
network mask	Subnet IP address.

Command Default

IP address trusted list is disabled.

Command Modes

Voice Service Voip.

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use this command to manually add unique and multiple IP addresses to a list of trusted IP addresses. You can add up to 100 IPv4 addresses in the ip address trusted list. No duplicate IP addresses are allowed.

Examples

The following is a sample output from this command displaying a list of trusted IP addresses:

```
Router #show ip address trusted list
IP Address Trusted Authentication
 Administration State: UP
 Operation State:      UP
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 Session Targets:
Peer Tag        Oper State      Session Target
--------        ----------      --------------
11              DOWN            ipv4:1.3.45.1
1               UP              ipv4:1.3.45.1
IP Address Trusted List:
 ipv4 172.19.245.1
 ipv4 172.19.247.1
 ipv4 172.19.243.1
 ipv4 171.19.245.1
 ipv4 171.19.10.1
```

Related Commands

Command	Description
IP address trusted authenticate	Enables IP address trusted authentication for incoming VoIP calls.
IP address trusted code-block cause	Allows to issues a cause-code when the incoming call is rejected by the IP address trusted authentication.

ip qos dscp (telephony-service and voice register global)

To set the Differentiated Services Code Point (DSCP) for marking the quality of service (QoS) requirements for each packet, use the **ip qos dscp** command in telephony-service or voice register global configuration mode. To reset to the default value, use the **no** form of this command.

ip qos dscp {*numberafcs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}
no ip qos dscp {*numberafcs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}

Syntax Description

number	DSCP value. Range: 0 to 63.
af	Sets DSCP to assured forwarding bit pattern. • **af11**—bit pattern 001010 • **af12**—bit pattern 001100 • **af13**— bit pattern 001110 • **af21**— bit pattern 010010 • **af22**— bit pattern 010100 • **af23**— bit pattern 010110 • **af31**— bit pattern 011010 • **af32**— bit pattern 011100 • **af33**— bit pattern 011110 • **af41**—bit pattern 100010 • **af42**—bit pattern 100100 • **af43**—bit pattern 100110
cs	Sets DSCP to class-selector codepoint. • **cs1**—codepoint 1 (precedence 1) • **cs2**—codepoint 2 (precedence 2) • **cs3**—codepoint 3 (precedence 3) • **cs4**—codepoint 4 (precedence 4) • **cs5**—codepoint 5 (precedence 5) • **cs6**—codepoint 6 (precedence 6) • **cs7**—codepoint 7 (precedence 7)
default	Sets DSCP to default bit pattern of 000000.
ef	Sets DSCP to expedited forwarding bit pattern 101110.
media	Applies DSCP to media payload packets.
service	Applies DSCP to phone service including HTTP traffic.
signaling	Applies DSCP to signaling packets.
video	Applies DSCP to video stream.

Command Default

DSCP for media is **ef**. DSCP for service is **0**. DSCP for signaling is **cs3**. DSCP for video is **af41**.

Command Modes

Telephony-service configuration (config-telephony)
Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command allows you to set different priority levels for different types of network traffic sent by the Cisco Unified CME router. Differentiated Services is a method of prioritizing specific network traffic based on the QoS specified by each packet. You can set different DSCP values, for example, for video and audio streams.

Cisco Unified CME downloads the configured DSCP value to the phones in their configuration files and all control messages and RTP streams are marked with the preferred DSCP value. Use this command in telephony-service mode to set the DSCP for SCCP phones. Use the command in voice register global mode to set the value for SIP phones.

If the DSCP is configured for the gateway interface using the **service-policy** command or in the dial peer using the **ip qos dscp** command, the value set with those commands takes precedence over the DSCP value configured with this command.

Examples

The following examples show the configuration of DSCP for different types of packets .

```
voice register global
 mode cme
 ip qos dscp af11 media
 ip qos dscp cs2 signal
 ip qos dscp af43 video
 ip qos dscp 25 service

telephony-service
 load 7960-7940 P00308000500
 max-ephones 100
 max-dn 240
 ip source-address 10.7.0.1 port 2000
 ip qos dscp af11 media
 ip qos dscp cs2 signal
 ip qos dscp af43 video
 ip qos dscp 25 service
```

Related Commands

Command	Description
ip qos dscp	Sets the DSCP for QoS in a dial peer.
service-policy	Assigns a policy map to an interface that will be used as the service policy for the interface.

ip source-address (credentials)

To enable the Cisco Unified CME or Cisco Unified SRST router to receive credential service messages through the specified IP address and port, use the **ip source-address** command in credentials configuration mode. To disable the router from receiving messages, use the **no** form of this command.

ip source-address *ip-address* [**port** [*port*]]
no ip source-address

Syntax Description

ip-address	Router IP address, typically one of the addresses of the Ethernet port of the local router.
port *port*	(Optional) TCP port for credentials service communication. Range is from 2000 to 9999. Cisco Unified CME default is 2444. SRST default is 2445.

Command Default

Default port number in Cisco Unified CME is 2444. Default port number in Cisco Unified SRST is 2445.

Command Modes

Credentials configuration (config-credentials)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco SRST 3.3	This command was introduced for Cisco SRST.
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced for Cisco Unified CME.
12.4(9)T	Cisco Unified CME 4.0	This command for Cisco Unified CME was integrated in Cisco IOS Release 12.4(9)T.

Usage Guidelines

Cisco Unified CME

This command is used with Cisco Unified CME phone authentication to identify a Cisco Unified CME router on which a CTL provider is being configured.

Cisco Unified SRST

The **ip source-address** command is a mandatory command to enable secure SRST. If the port number is not provided, the default value (2445) is used. The IP address is usually the IP address of the secure SRST router.

Examples

Cisco Unified CME

The following example creates a CTL provider on a Cisco Unified CME router that is not running the CTL client.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint ctlpv
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```

Cisco Unified SRST

The following example enters credentials configuration mode and sets the IP source address and port:

```
Router(config)# credentials
Router(config-credentials)# ip source-address 10.6.21.4 port 2445
```

Related Commands

Command	Description
ctl-service admin	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
debug credentials	Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider and the CTL client or between an SRST router and Cisco Unified CallManager.
show credentials	Displays the credentials settings on a Cisco Unified CME or SRST router.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.

ip source-address (telephony-service)

To identify the IP address and port through which IP phones communicate with a Cisco Unified CME router, use the **ip source-address** command in telephony-service or group configuration mode. To disable the router from receiving messages from Cisco Unified IP phones, use the **no** form of this command.

ip **{** *ipv4_address*| *ipv6_address***}** **[port***port***]** **[secondary** **{***ipv4 address* |*ipv6 address***}** **[rehome***seconds***]]**
[any-match | strict-match]
no ip source-address

Syntax Description

ipv4_address	IPv4 address of the router, typically one of the addresses of the Ethernet port of the router.
ipv6_address	In Cisco Unified CME 8.0 and later versions: IPv6 address of the router, typically one of the addresses of the Ethernet port of the router.
port *port*	(Optional) TCP/IP port number to use for Skinny Client Control Protocol (SCCP). Default is 2000. For IPv4 only: Range is from 2000 to 9999. **Note** For IPv6, do not configure the port number to change from the default value (2000).
secondary	(Optional) Second Cisco Unified CME router with which phones can register if the primary Cisco Unified CME router fails. **Note** For dual-stack (IPv4 and IPv6) mode: Only an IPv4 address can be configured for a secondary router.
rehome *seconds*	(Optional) Used only by Cisco Unified IP phones that have registered with a Cisco Unified Survivable Remote Site Telephony (SRST) router. This keyword defines a delay that is used by phones to verify the stability of their primary SCCP controller (Cisco Unified Communications Manager or Cisco Unified CME) before the phones reregister with it. This parameter is ignored by phones unless they are registered to a secondary Cisco Unified SRST router. The range is from 0 to 65535 seconds. The default is 120 seconds. The use of this parameter is a phone behavior and is subject to change, based on the phone type and phone firmware version.
strict-match	(Optional) Requires strict IP address checking for registration.

Command Default

The IP address for communicating with phones is not defined.

Command Modes

Telephony-service configuration (config-telephony)
Group configuration (conf-tele-group)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	The **secondary** *ip-address* and **rehome** *seconds* keyword-argument pairs were added.
12.4(9)T	Cisco Unified CME 4.0	The **secondary** *ip-address* and **rehome** *seconds* keyword-argument pairs were added.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was added to VRF group mode.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. Support for IPv6 was added and the *ipv4-address* and ipv6-*address arguments replaced the generic ip-address argument.*
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command enables a router to receive messages from Cisco Unified IP phones through the specified IP address and port.

The Cisco Unified CME router cannot communicate with Cisco Unified CME phones if the IP address of the port to which they are attached is not configured. In Cisco Unified CME 8.0 and later versions, the Cisco Unified CME router can receive messages from IPv6-enabled or IPv4-enabled IP phones or from phones in dual-stack (both IPv6 and IPv4) mode.

- In Cisco Unified CME 8.0 and later versions: If the IP phones connected to Cisco Unified CME were configured for dual-stack mode by using **dual-stack** keyword with the **protocol mode** command, configure this command with the IPv6 address.
- In Cisco Unified CME 8.0 and later versions: If the IP phones to be connected to the port to be configured are IPv4-enabled only *or* IPv6-enabled only, configure this command with the corresponding IPv4 or IPv6 address.

For IPv6: Do not configure the **port** *port* keyword argument combination in this command to change the value from the default (2000). If you change the port number, IPv6 CEF packet switching engine will not be able to handle the IPv6 SCCP phones and various packet handling problems may occur when more than a dozen (approximately) calls in IPv6 are going on.

Use the **strict-match** keyword to instruct the router to reject IP phone registration attempts if the IP server address used by the phone does not match the source address.

Prior to Cisco IOS Telephony Services (Cisco ITS) V2.1, this command helped the router to autogenerate the SEPDEFAULT.cnf file, which was stored in the flash memory of the router. The SEPDEFAULT.cnf file contains the IP address of one of the Ethernet ports of the router to which the phone should register.

In ITS V2.1 and in Cisco CME 3.0 and later versions, the configuration files were moved to system:/its/. The file named Flash:SEPDEFAULT.cnf that was used with previous Cisco ITS versions is obsolete, but is retained as system:/its/SEPDEFAULT.cnf to support upgrades from older phone firmware.

For systems using Cisco ITS V2.1 or later versions, the IP phones receive their initial configuration information and phone firmware from the TFTP server associated with the router. In most cases, the phones obtain the IP address of their TFTP server using the **option 150** command and Dynamic Host Configuration Protocol (DHCP). For Cisco ITS or Cisco CME operation, the TFTP server address obtained by the Cisco Unified IP phones should point to the router IP address. The Cisco IP phones attempt to transfer a configuration file called XmlDefault.cnf.xml. This file is automatically generated by the router through the **ip source-address** command and is placed in router memory. The XmlDefault.cnf.xml file contains the IP address that the phones use to register for service, using the SCCP. This IP address should correspond to a valid Cisco CME router IP address (and may be the same as the router TFTP server address).

Similarly, when an analog telephone adapter (ATA) such as the ATA-186 is attached to the Cisco Unified CME router, the ATA receives very basic configuration information and firmware from the TFTP server XmlDefault.cnf.xml file. The XmlDefault.cnf.xml file is automatically generated by the Cisco Unified CME router with the **ip source-address** command and is placed in the router's flash memory.

By specifying a second Cisco Unified CME router in the **ip source-address** command, you improve the failover time for phones.

Examples

The following example sets the IP source address and port:

```
Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.6.21.4 port 2000 strict-match
```

The following example establishes the router at 10.5.2.78 as a secondary router:

```
Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.0.0.1 port 2000 secondary 10.5.2.78
```

Cisco Unified CME 8.0 and Later Versions

The following example shows how to configure this command with an IPv6 address. Do not change the port number from the default value (2000) when you configure an IPv6 address.

```
Router(config)# telephony-service
Router(config-telephony)#protocol mode ipv6
Rounter(config-telephony)ip source-address 2001:10:10:10::3
```

The following example shows how to configure an IP address for dual-stack mode. When the IP phones are configured for dual-stack mode, the IP address of the router port to which the IP phones are connected must be an IPv6 address. For dual-stack mode, the address of the secondary router must be an IPv4 address.

```
Router(config)# telephony-service
Router(config-telephony)# protocol mode dual-stack
Router(config-telephony)# ip source address
2001:10:10:10::3 secondary 10.5.2.78
Router(config-telephony)#
```

Related Commands

Command	Description
option	Configures DHCP server options.
protocol mode	Configures a preferred IP-address mode for SCCP IP phones in Cisco Unified CME.

Cisco Unified CME Commands: K

keepalive (ephone and ephone-template)

To set the length of the time interval between successive keepalive messages from the Cisco Unified CME router to a particular IP phone, use the **keepalive** command in ephone or ephone-template configuration mode. To reset this length to the default value, use the **no** form of this command.

keepalive *seconds*
no keepalive

Syntax Description

seconds	Interval time, in seconds. Range is from 10 to 65535. Default is 30.

Command Default

Default is 30 seconds

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)T	Cisco CME 2.1	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command allows the keepalive interval to be set for individual phones, typically so that wireless phone batteries are not run down too quickly by overly frequent keepalive signals.

If the router fails to receive three successive keepalive messages, it considers the phone to be out of service until the phone reregisters.

If the **keepalive (telephony-service)** command and this command are set to different time intervals, the value that you set in ephone configuration mode has priority for the particular phone only.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example sets the keepalive interval to 300 seconds:

```
Router(config)# ephone 1
Router(config-ephone)# keepalive 300
```

Related Commands

Command	Description
ephone-template (ephone)	Applies template to ephone being configured.

Command	Description
keepalive (telephony-service)	Sets the time interval for keepalive messages between IP phones and the Cisco Unified CME router.

keepalive (telephony-service)

To set the length of the time interval between successive keepalive messages from the Cisco CallManager Express router to IP phones, use the **keepalive** command in telephony-service configuration mode. To reset this length to the default value, use the **no** form of this command.

keepalive *seconds*
no keepalive

Syntax Description

seconds	Interval time, in seconds. Range is from 10 to 65535. Default is 30.

Command Default

Default is 30 seconds.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

If the router fails to receive three successive keepalive messages, it considers the phone to be out of service until the phone reregisters.

If the **keepalive (telephony-service)** command and the **keepalive (ephone)** command are set to different time intervals, the value that you set in ephone configuration mode has priority for the particular phone only.

Examples

The following example sets the keepalive time interval to 40 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# keepalive 40
```

Related Commands

Command	Description
keepalive (ephone)	Sets the time interval for keepalive messages between a particular IP phone and the Cisco CME router.

keepalive (voice register global)

To set the length of time interval between successive keepalive messages from SIP phones to the Cisco Unified CME router, use the **keepalive** command in voice register global configuration mode. To reset this timer duration to the default value, use the **no** form of this command.

keepalive *seconds*
no keepalive

Syntax Description

seconds	Sets the time interval, in seconds, between keepalive messages that are sent to the router by SIP Phones. If the interval is set to a larger value, it is possible for notification to be delayed when the primary router goes down. Range is from 120 to 65535. Default is 120 seconds.

Command Default

Default is 120 seconds.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1	Cisco Unified CME 11.6	This command was introduced.

Usage Guidelines

If the primary router fails, a SIP phone will not receive an acknowledgment (200 OK) to its REGISTER message to the primary router, and it will immediately failover to the secondary Cisco Unified CME router.

Examples

The following example sets the keepalive time interval to 200 seconds:

```
Router(config)# voice register global
Router(config-register-global)# keepalive 200
```

Related Commands

Command	Description
keepalive (telephony-service)	Sets the length of the time interval between successive keepalive messages from the Cisco Unified CME router to SCCP phones.

keepalive (voice register session-server)

To define the duration for registrations of external feature servers after which the registration expires, use the **keepalive** command in voice register session-server configuration mode. To return to the default, use the **no** form of this command.

keepalive *seconds*
no keepalive

Syntax Description

seconds	Duration for registration, in seconds. Range: 60 to 3600. Default: 300.

Command Default

Default is 300 seconds.

Command Modes

Voice register session-server configuration (config-register-fs)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW2	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4920)T.

Usage Guidelines

This command defines the duration for registration, in seconds, after which the registration expires unless the feature server reregisters before the registration expiry.

Examples

The following partial output shows the configuration for a session manager for an external feature server, including a keepalive expiry of 360 seconds:

```
router# show running-configuration

!
!
voice register session-server 1
 register-id CSR1
 keepalive 360
```

Related Commands

Command	Description
register id	Creates an ID for explicitly identifying an external feature server during Register requests.

keepalive (vpn-profile)

To specify the duration of time required to generate a keepalive message to the VPN concentrator, use the **keepalive** command in vpn-profile configuration mode.

keepalive seconds

Syntax Description

seconds	Duration for a vpn-profile session, in seconds. Range: 0 to 120. Default: 60.

Command Default

Default is 60 seconds.

Command Modes

Vpn-profile configuration (conf-vpn-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to specify the amount of time required to generate a keepalive message to the VPN concentrator. The keepalive session ranges from 0 to 120 seconds. The default keepalive session is 60 seconds.

Examples

The following example shows the keepalive duration set to 50 seconds for vpn-profile 1.

```
Router#show run
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  keepalive 50
  host-id-check disable
 vpn-profile 2
  mtu 1300
  password-persistent enable
  host-id-check enable
 sip
!
voice class media 10
 media flow-around
!
```

Related Commands

Command	Description
vpn-profile	Defines a VPN-profile.

keep-conference

To allow conference initiators to exit from conference calls and to either end or maintain the conference for the remaining parties, use the **keep-conference** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

keep-conference [**drop-last**] [**endcall**] [**local-only**]
no keep-conference

Syntax Description

drop-last	(Optional) The action of the Confrn soft key is changed; the conference initiator can press the Confrn soft key (IP phone) or hookflash (analog phone) to drop the last party. **Note** Analog phones connected to the Cisco Unified CME system through a Cisco VG 224 require Cisco IOS Release 12.3(11)YL1 or a later release to use this feature.
endcall	(Optional) The action of the EndCall soft key is changed; the conference initiator can hang up or press the EndCall soft key to leave the conference and keep the other two parties connected. **Note** If this option is not enabled, pressing the EndCall soft key terminates the conference and disconnects all parties.
local-only	(Optional) The conference initiator can hang up to end the conference and leave the other two parties connected only if one of the remaining parties is local to the Cisco Unified CME system (an internal extension).

Command Default

A conference initiator can hang up or press the EndCall soft key to end a conference and disconnect all parties or press the Confrn soft key to drop only the last party that was connected to the conference.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **drop-last** and **local-only** keywords were added, and this command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	The **drop-last** and **local-only** keywords, and this command in ephone-template configuration mode were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Note This feature uses call transfer to connect the two remaining parties of a conference when a conference initiator leaves the conference. To use this feature, you must configure the **transfer-system** command using the **full-blind**, **full-consult**, or **full-consult dss** keywords.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

If the **keep-conference** command is configured with no keywords, a conference initiator can hang up to leave the conference and the other two parties will remain connected. Alternatively, the conference initiator can use the EndCall soft key to terminate the conference and disconnect all parties.

If the **keep-conference** command is configured with no keywords, a conference initiator can use the Confrn soft key (IP phone) or hookflash (analog phone) to break up the conference but stay connected to both parties. The oldest call will be put on hold, and the most recent call will be actively connected to the initiator. The conference initiator can navigate between the two parties by pressing the Hold soft key or the appropriate line button on the phone.

If the **endcall** keyword is used, the conference initiator can hang up or press the EndCall soft key to leave the conference with the other two parties remaining connected.

In Cisco CME 3.2.3 and later versions, if the **keep-conference** command is not configured (the default) or if the **no keep-conference** command is used, a conference initiator can drop the last party that was added to the conference by pressing the Confrn soft key (IP phone) or hookflash (analog phone).

Note Analog phones connected to the Cisco Unified CME system through a Cisco VG 224 require Cisco IOS Release 12.3(11)YL1 or a later release to use this feature.

Examples

In the following example, extension 3555 initiates a three-way conference. After the conference is established, extension 3555 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. If extension 3555 hangs up from the conference, the other two parties remain connected if one of them is local to the Cisco Unified CME system.

```
ephone-dn 35
 number 3555
ephone 24
 button 1:35
 keep-conference drop-last local-only
```

In the following example, extension 3666 initiates a three-way conference. After the conference is established, extension 3666 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. Also, extension 3666 can hang up from a three-way conference to terminate the conference and disconnect all parties or can press the EndCall soft key to leave the conference and keep the other two parties connected.

```
ephone-dn 36
 number 3666
ephone 25
 button 1:36
 keep-conference drop-last endcall
```

In the following example, extension 3777 initiates a three-way conference. After the conference is established, extension 3777 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. Also, extension 3777 can hang up from a three-way conference to terminate the conference and disconnect all parties or press the

EndCall soft key to leave the conference and keep the other two parties connected only if one of the two parties is local to the Cisco Unified CME system.

```
ephone-dn 38
 number 3777
ephone 27
 button 1:38
 keep-conference drop-last endcall local-only
```

In the following example, extension 3999 initiates a three-way conference. After the conference is established, extension 3999 can hang up to terminate the conference and disconnect all parties or press the EndCall soft key to leave the conference and keep the other two parties connected only if one of the two parties is local to the Cisco Unified CME system.

```
ephone-dn 39
 number 3999
ephone 29
 button 1:39
 keep-conference endcall local-only
```

Related Commands

Command	Description
ephone-template (ephone)	Applies template to ephone being configured.
max-conferences	Sets the maximum number of three-party conferences simultaneously supported by the Cisco Unified CME router.
transfer-system	Specifies the call transfer method for IP phone extensions that use the ITU-T H.450.2 standard.

keep-conference (voice register)

To allow IP phone conference initiators to exit from conference calls and keep the remaining parties connected, use the **keep-conference** command in voice register pool configuration mode or voice register template configuration mode. To disable the keep-conference feature, use the **no** form of this command.

keep-conference
no keep-conference

Syntax Description

This command has no arguments or keywords.

Command Default

Default is enabled.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
Cisco IOS XE Everest 16.5.1b	Unified CME 11.7	This command was supported in voice register template configuration mode.

Usage Guidelines

When the conference initiator hangs up, Cisco Unified Communications Manager Express (Cisco Unified CME) executes a call transfer to connect the two remaining lines. The remaining calls are transferred without consultation. To facilitate call transfer, the **transfer-attended** command or **transfer-blind** command must be enabled.

Conference initiators can disconnect from their conference calls by pressing the Confrn (conference) soft key. When an initiator uses the Confrn soft key to disconnect from the conference call, the oldest call leg is put on hold, leaving the initiator connected to the most recent call leg. The conference initiator can then navigate between the two separate parties by pressing either the Hold soft key or the line buttons to select the desired call.

Examples

The following example shows how to configure this command, if it was previously disabled, to keep remaining conference legs after the conference initiator hangs up.

```
Router(config)# voice register pool 1
Router(config-register-pool)# keep-conference
```

The following example shows how to configure this command under voice register template configuration mode.

```
Router(config)# voice register template 1
Router(config-register-template)# keep-conference
```

Related Commands

Command	Description
conference (voice register template)	Enables a soft key for conference in a SIP phone template.
max-conferences	Sets the maximum number of three-party conferences simultaneously supported by the Cisco CME router.
transfer-attended (voice register template)	Enables a soft key for attended transfer in a SIP phone template.
transfer-blind (voice register template)	Enables a soft key for blind transfer in a SIP phone template.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

keygen-retry

To specify the number of times that a CAPF server sends a key-generation request, use the **keygen-retry** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

keygen-retry *number*
no keygen-retry

Syntax Description

number	Number of retries. Range is from 0 to 100. Default is 3.

Command Default

Number of retries is 3.

Command Modes

CAPF-server configuration (config-capf-server)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example specifies that the key generation process should be tried 5 times.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

Related Commands

Command	Description
keygen-timeout	Specifies the number of minutes that the CAPF server waits for a key-generation response from a phone.

keypad-normalize

To impose a 200-millisecond delay before each keypad message from an IP phone, use the **keypad-normalize** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

keypad-normalize
no keypad-normalize

Syntax Description

This command has no keywords or arguments.

Command Default

Keypad messages are handled as fast as the system can handle them, without an imposed delay.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command normalizes the processing of incoming keypad messages from an IP phone so that one message is processed every 200 milliseconds. This is useful for handling the personal speed dial (fastdial) feature when the destination of the call tends to be slower in accepting the digits, or when converting keypad messages into appropriate digit events on the network side, such as RFC 2833 digits.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example normalizes the sending of digits from ephone 43.

```
ephone 43
 button 1:29
 keypad-normalize
```

Related Commands

Command	Description
ephone-template (ephone)	Applies template to ephone being configured.

keyphone

To designate a Cisco Unified IP phone as a marked or "key" phone when using the Cisco Unified CME eXtensible Markup Language (XML) application program interface (API), use the **keyphone** command in ephone or ephone-template configuration mode. To remove the keyphone designation, use the **no** form of this command.

keyphone
no keyphone

Syntax Description

This command has no arguments or keywords.

Command Default

The phone that is being configured is not a "key" phone.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with the XML API to mark a Cisco Unified IP phone as a "key" phone to be tracked while using the XML API. The XML API can be instructed to report the status of only the "key" phones in the system for network management purposes, for example.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example sets the phone with the phone tag of 1 as a "key" phone for the XML API:

```
Router(config)# ephone 1
Router(config-ephone)# keyphone
```

Related Commands

Command	Description
ephone-template (ephone)	Applies template to ephone being configured.

Cisco Unified CME Commands: L

label

To create a text identifier instead of a phone-number display for an extension on an IP phone console, use the **label command in** ephone-dn configuration mode. To delete a label, use the **no** form of this command.

label *string*
no label *string*

Syntax Description

string	Alphanumeric string of up to 30 characters.

Command Default

No label is defined.

Command Modes

Ephone-dn configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

One label is allowed per extension (ephone-dn). The ephone-dn must already have a number that was set using the **number** command before a label can be created for it.

This command must be followed by a quick reboot of the phone on which the label appears, using the **restart** command.

Examples

The following example creates three phone labels to appear in place of three phone numbers on IP phone console displays:

```
Router(config)# ephone-dn 10
Router(config-ephone-dn)# label user10
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 20
Router(config-ephone-dn)# label user20
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 30
Router(config-ephone-dn)# label user30
Router(config-ephone-dn)# exit
```

Related Commands

Command	Description
number	Associates a telephone or extension number with an ephone-dn in a Cisco CME system.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

label (voice register dn)

To create a text identifier instead of a phone-number display for an extension on a SIP phone console, use the **label command in** voice register dn configuration mode. To delete a label, use the **no** form of this command.

label string
no label string

Syntax Description

string	Alphanumeric string of up to 30 characters.

Command Default

No label is created.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

One label is allowed per extension (directory number). The directory number must already have a number that was set by using the **number** command before a label can be created for it.

After you configure this command, restart the phone by using the **reset** command.

Examples

The following example shows how to create three phone labels to appear in place of three phone numbers on Cisco IP phone console displays:

```
Router(config)#  voice register dn 10
Router(config-register-dn)# label user10
Router(config-register-dn)# exit
Router(config)# voice register dn 20
Router(config-register-dn)# label user20
Router(config-register-dn)# exit
Router(config)# voice register dn 30
Router(config-register-dn)# label user30
Router(config-register-dn)# exit
```

Related Commands

Command	Description
number (voice register dn)	Associates a telephone or extension number with a directory number in a Cisco CME system.
reset (voice register pool)	Performs a compete reboot of a single SIP phone associated with a Cisco CME router.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.

list (ephone-hunt)

To create a list of extensions that are members of a Cisco Unified CME ephone hunt group, use the **list** command in ephone-hunt configuration mode. To remove a list from the router configuration, use the **no** form of this command.

list*number***[,***number***...]**
no list

Syntax Description

number	Preconfigured extension or E.164 number. An asterisk (*) can take the place of an extension number to represent a wildcard slot. An agent at an authorized ephone-dn can dynamically join and leave a hunt group if a wildcard slot is available. There can be up to 20 wildcard slots in a hunt group.

Command Default

No list is defined.

Command Modes

Ephone-hunt configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)XL	Cisco CME 3.2.1	The number of ephone-dns allowed was increased to 20.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The use of wildcard asterisks (*) in the *dn-number* argument was introduced.
12.4(9)T	Cisco Unified CME 4.0	The use of wildcard asterisks (*) in the *dn-number* argument was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to create a list of member numbers for defining a hunt group.

List must contain 1 to 20 numbers.

A number cannot be added to a list unless it was already defined by using the **number** command.

Add or delete all numbers in a hunt group list at one time. You cannot add or single number to an existing list or remove one number from a list.

Any number in the list cannot be a pilot number of a parallel hunt group.

To allow dynamic membership in a hunt group, use asterisks to represent wildcard slots in the **list** command. To allow an ephone-dn to use one of the wildcard slots to dynamically join a hunt group, use the **ephone-hunt login** command under that ephone-dn. Ephone-dns are disallowed from joining hunt groups by default, so you have to explicitly allow this behavior for each ephone-dn that you want to be able to log into hunt groups.

The **show ephone-hunt** command displays the numbers associated to ephone-dns that are joined to groups at the time that the command is run, in addition to static members of the hunt group. Static hunt group members are the numbers that are explicitly named in the **list** command.

Examples

The following example creates sequential hunt group number 7, which contains four static members (ephone-dns):

```
Router(config)# ephone-hunt 7 sequential
Router(config-ephone-hunt)# list 7711, 7712, 7713, 7714
```

The following example creates five ephone-dns and a hunt group that includes the first two ephone-dns as static members and two wildcard slots for dynamic hunt group members. The last three ephone-dns are enabled for dynamic membership in the hunt group. Any of them can join the hunt group whenever one of the wildcard slots is available. Once an ephone-dn has joined a hunt group, it can leave at any time.

```
ephone-dn 22
 number 4566
ephone-dn 23
 number 4567
ephone-dn 24
 number 4568
 ephone-hunt login
ephone-dn 25
 number 4569
 ephone-hunt login
ephone-dn 26
 number 4570
 ephone-hunt login
ephone-hunt 1 peer
 list 4566,4567,*,*
 final 7777
```

Related Commands

Command	Description
ephone-hunt login	Allows an ephone-dn to dynamically join and leave an ephone hunt group.
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
max-redirect	Changes the current number of allowable redirects in a Cisco CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.
number (ephone-dn)	Associates an extension or telephone number with a directory number.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.

Command	Description
show ephone-hunt	Displays ephone-hunt group configuration, current status, and statistics.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

list (voice hunt-group)

To define a list of extensions that are members of a voice hunt-group, use the **list** command in voice hunt-group configuration mode. To remove a list, use the **no** form of this command.

list*number, number***[,***number***...]**
no list

Syntax Description

number	Extension or E.164 number assigned to a phone in Cisco Unified CME. List must contain 2 to 32 numbers.

Command Default

By default, hunt group list is not defined.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	The maximum numbers allowed in a list was expanded from 10 to 32 and support for SCCP phones was added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.4(3)M	Cisco Unified CME 10.5	This command was modified to include support for wildcards which is indicated by "*" . symbol.

Usage Guidelines

This command creates the list of numbers to include in a voice hunt-group. Phones with these numbers ring when the hunt group pilot number is dialed. The numbers must be assigned to directory numbers with the **number** command.

All numbers in a hunt group list are added or deleted at one time. You cannot add a number to an existing list or remove a number from a list.

The pilot number of a parallel hunt group and shared-line numbers are not supported.

A phone number associated with an FXO port is not supported in parallel hunt groups.

Examples

The following example shows how to create a sequential hunt group containing four extensions and a wildcard slot:

```
Router(config)# voice hunt-group 1 sequential
Router(config-voice-hunt-group)# list 7711, 7712, 7713, 7714, *
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.

Command	Description
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next phone number in a peer hunt-group list before proceeding to the final number.
number (ephone-dn)	Associates an extension or telephone number with a directory number.
number (voice register dn)	Associates an extension or telephone number with a directory number.
pilot (voice hunt-group)	Defines the phone number that callers dial to reach a voice hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last number in the hunt group.

live-record

To define the extension number that is dialed when the LiveRcd soft key is pressed on a Cisco Unified IP Phone, use the **live-record** command in telephony-service configuration mode. To reset to the default value, use the **no** form of this command.

live-record *phone-number*
no live-record

Syntax Description

phone-number	Telephone number that is dialed when the LiveRcd soft key is pressed.

Command Default

Live record is disabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies the telephone number that is speed-dialed to access the Live Record feature when the LiveRcd soft key on a Cisco Unified IP phone is pressed. This telephone number is used for all Cisco Unified IP phones connected to the router.

This telephone number must match the Live Record number configured in Cisco Unity Express.

Examples

The following example shows that the phone number 914085550100 is speed-dialed to record a call when the LiveRcd button is pressed:

```
Router(config)# telephony-service
Router(config-telephony)# live-record 914085550100
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an ephone.
softkeys connected	Modifies the order and type of soft keys that display on an IP phone during the connected call state.
voicemail	Defines the telephone number that is speed-dialed when the Messages button is pressed on an IP phone.

load (telephony-service)

To associate a type of Cisco Unified IP phone with a phone firmware file, use the **load** command in telephony-service configuration mode. To disassociate a type of phone from a phone firmware file, use the **no** form of this command.

load *phone-type* *firmware-file*
no load *phone-type*

Syntax Description

phone-type	Type of phone. The following phone types are predefined in the system: • **6945**—Cisco Unified IP Phone 6945. • **7902**—Cisco Unified IP Phone 7902G. • **7905**—Cisco Unified IP Phone 7905G. • **7910**—Cisco Unified IP Phone 7910 and 7910G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco Unified IP Phone 7912G. • **7914**—Cisco Unified IP Phone 7914 Expansion Module. • **7920**—Cisco Unified Wireless IP Phone 7920. • **7921**—Cisco Unified Wireless IP Phone 7921. • **7931**—Cisco Unified IP Phone 7931G. • **7935**—Cisco Unified IP Conference Station 7935. • **7936**—Cisco Unified IP Conference Station 7936. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941G-GE. • **7942**—Cisco Unified IP Phone 7942. • **7945**—Cisco Unified IP Phone 7945 • **7960-7940**—Cisco Unified IP Phones 7960 and 7960G and Cisco Unified IP Phone 7940 and 7940G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961G-GE. • **7962**—Cisco Unified IP Phone 7962. • **7965**—Cisco Unified IP Phone 7965. • **7970**—Cisco Unified IP Phone 7970G. • **7971**—Cisco Unified IP Phone 7971G-GE. • **7975**—Cisco Unified IP Phone 7975. • **7985**—Cisco Unified IP Phone 7985. • **8941**—Cisco Unified IP Phone 8941. • **8945**—Cisco Unified IP Phone 8945. • **ata**—Cisco ATA-186 and Cisco ATA-188. **Note** You can also add a new phone type to your configuration by using the **ephone-type** command.

firmware-file	Filename of the IP phone firmware for a particular phone type. • In Cisco Unified CME 7.0/4.3 and earlier versions, do not use the file suffix (.bin, .sbin, .loads) for any phone type except the Cisco ATA and Cisco Unified IP Phone 7905 and 7912. • In Cisco Unified CME 7.0(1) and later versions, you must use the complete filename, including the file suffix, for phone firmware versions later than version 8-2-2 for all phone types. • Filenames are case sensitive.

Command Default

Firmware files are not associated with phone types.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(11)YT	Cisco ITS 2.1	Support was added for the Cisco IP Phone 7914 Expansion Module.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The following keywords were added to this command: **7902**, **7905**, and **7912**.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2.1	The **7970** keyword was added.
12.3(14)T	Cisco CME 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were added.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	The **7931** keyword was added.
12.4(4)XC4	Cisco Unified CME 4.0(3)	The **7931** keyword was added.
12.4(11)T	Cisco Unified CME 4.0(3)	This command was integrated into Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **7921** and **7985** keywords were introduced.

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(15)T1	Cisco Unified CME 4.1(1)	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were introduced.
12.4(15)XZ	Cisco Unified CME 4.3	Support for user-defined phone types created with the **ephone-type** command was added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
12.4(20)YA	Cisco Unified CME 7.0(1)	Support for automatically creating bindings for firmware files only if the cnf-file location is flash or slot0 was added.
12.4(20)T1	Cisco Unified CME 7.0	The **7925** keyword was introduced.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was integrated into Cisco IOS Release 12.4(22)T.
15.2(1)T	Cisco Unified CME 8.8	This command was modified. The **6945**, **8941**, and **8945** keywords were added.

Usage Guidelines

This command updates the Cisco Unified CME configuration file for the specified type of Cisco Unified IP phone to add the name of the firmware file to be loaded by a particular phone type. The firmware filename also provides the version number for the phone firmware that is in the file. When a phone is started up or rebooted, the phone reads the configuration file to determine which firmware file it must load and then looks for that firmware file on the TFTP server.

If applicable, Cisco Unified IP phones update themselves with new phone firmware whenever they are started up or rebooted.

A separate **load** command is needed for each type of phone. The Cisco Unified IP Phones 7940 and 7940G and the Cisco Unified IP Phones 7960 and 7960G have the same phone firmware and share the **7960-7940** keyword.

Before Cisco Unified CME 7.0(1):

- Do not include the file suffix (.bin, .sbin, .loads) for any phone type except Cisco ATA and Cisco Unified IP Phone 7905 and 7912 when you configure the **load** command in telephony-service configuration mode. For example:

```
Router(config-telephony)# load 7941 SCCP41.8-2-2SR2S
Router(config-telephony)#
```

- You must also configure the **tftp-server** command to enable TFTP access to the firmware files by Cisco Unified IP phones.

In Cisco Unified CME 7.0(1) and later versions:

- When specifying the load command for phone firmware versions later than version 8-2-2 for all phone types and you use the file suffix in the filename, the tftp-server bindings are automatically added for all the files forwarded for that load. For example:

```
Router(config-telephony)# load 7941 SCCP41.8-3-3S.loads
Router(config-telephony)#
```

- The **load** command is enhanced to automatically create TFTP bindings for phone firmware files if the **cnf-file location** command is configured with the **flash** or **slot0** keyword. You are no longer required to configure the **tftp-server** command to create TFTP bindings only if the location of the cnf files is router flash or slot 0 memory. If the **cnf-file location** command is configured for something other than flash or slot 0, such as a TFTP server (url) or system memory (system:its/), you must still configure the **tftp-server** command to create TFTP bindings for phone firmware files. Use the complete filename, including the file suffix, when you configure the **tftp-server** command for phone firmware versions later than version 8-2-2 for all phone types.

To verify TFTP bindings, including the dictionary, language, and tone configuration files that are associated with the ISO-3166 codes that have been selected, use the **show telephony-service tftp-bindings** command.

After associating a firmware file with a Cisco Unified IP phone, use the **reset** command to reboot the phone.

Examples

Cisco Unified CME 7.0 and Earlier Versions

The following example shows how to identify the Cisco Unified IP phone firmware file to be used by the Cisco Unified IP Phones 7960 and 7960G and Cisco Unified IP Phone 7910G:

```
Router(config)# telephony-service
Router(config-telephony)# load 7960-7940 P00303020209
Router(config-telephony)# load 7910 P00403020209
Router(config-telephony)# exit
Router(config)# tftp-server flash:P00303020209.bin
Router(config)# tftp-server flash:P00403020209.bin
```

Related Commands

Command	Description
cnf-file location	Specifies a storage location for phone configuration files.
ephone-type	Adds a Cisco Unified IP phone type by defining a phone-type template.
reset	Resets a Cisco Unified IP phone.
show telephony-service tftp-bindings	Provides a list of configuration files that are accessible to IP phones using TFTP.
tftp-server	Enables TFTP access to firmware files on the TFTP server.

load (voice register global)

To associate a type of IP phone with a phone firmware file, use the **load** command in voice register global configuration mode. To disassociate a type of phone from a phone firmware file, use the **no** form of this command.

load *phone-type firmware-file*
no load *phone-type firmware-file*

Syntax Description

phone-type	Type of IP phone. The following choices are valid: • **3905**—Cisco Unified IP Phone 3905. • **3951**—Cisco Unified IP Phone 3911 and 3951. • **6901**—Cisco Unified IP Phone 6901. • **6911**—Cisco Unified IP Phone 6911. • **6921**—Cisco Unified IP Phone 6921. • **6941**—Cisco Unified IP Phone 6941. • **6945**—Cisco Unified IP Phone 6945. • **6961**—Cisco Unified IP Phone 6961. • **7821**—Cisco Unified IP Phone 7821. • **7841**—Cisco Unified IP Phone 7841. • **7861**—Cisco Unified IP Phone 7861. • **7905**—Cisco Unified IP Phone 7905 and 7905G. • **7906**—Cisco Unified IP Phone 7906G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco Unified IP Phone 7912 and 7912G. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941GE. • **7942**—Cisco Unified IP Phone 7942. • **7945**—Cisco Unified IP Phone 7945. • **7960–7940**—Cisco Unified IP Phones 7940 and 7940G and Cisco IP Phones 7960 and 7960G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961GE. • **7962**—Cisco Unified IP Phone 7962. • **7965**—Cisco Unified IP Phone 7965. • **7970**—Cisco Unified IP Phone 7970G. • **7971**—Cisco Unified IP Phone 7971GE. • **7975**—Cisco Unified IP Phone 7975. • **ATA**—Cisco ATA-186 and Cisco ATA-188. • **ATA-187**—Cisco ATA-187. • **DX650**—Cisco DX650.
firmware-file	Filename for the Cisco Unified IP phone firmware to be associated with the IP phone type. Do not use the .bin or .load file extension, except for the Cisco Unified IP phone 7905, 7912, or ATA. Filenames are case sensitive.

Command Default

The firmware file is not associated with the type of phone.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970,** and **7971** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970**, and **7971** keywords were integrated into Cisco IOS Software Release 12.4(15)T.
12.4(15)XZ	Cisco Unified CME 4.3	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.2(1)T	Cisco Unified CME 8.8	This command was modified. The **3905** keyword was added.
15.2(2)T	Cisco Unified CME 9.0	This command was modified. The **6901**, **6911**, **6921**, **6941**, **6945**, **6961**, and **ATA-187** keywords were added.
15.4(3)M	Cisco Unified CME 10.5	This command was modified to provide support for Cisco Unified 7821, 7841, 7861 and DX650 IP phones.

Usage Guidelines

This command updates the Cisco Unified CME configuration file for the specified type of IP phone to add the name of the correct firmware file that the phone should load. This filename also provides the version number for the phone firmware that is in the file. Later, whenever a phone is started up or rebooted, the phone reads the configuration file to determine the name of the firmware file that it should load and then looks for that firmware file on the TFTP server.

A separate **load** command is needed for each type of phone. The Cisco Unified IP Phone 7940 and 7940G and Cisco Unified IP Phone 7960 and 7960G have the same phone firmware and share the **7960-7940** keyword. The Cisco Unified IP Phone 3911 and Cisco Unified IP Phone 3951 have the same phone firmware and share the **3951** keyword.

For certain IP phones, such as the Cisco Unified IP Phone 7906G, 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971G, there are multiple firmware files. For these phones, use the TERMnn.x-y-x-w.loads or SIPnn.x-y-x-w.loads firmware filename for the **load** command, without the .loads file extension. For these phones, you do not configure the **load** command for any firmware file other than the TERM.loads or SIP.loads firmware file.

Following the **load** command, use the **tftp-server** command to enable TFTP access to the file by Cisco Unified IP phones. The file extension is required when using the **tftp-server** command.

The **load** command must be followed by a reboot of the phones. Plug in a new IP phone or use the **reset** command to reboot an IP phone that is already connected to the Cisco router.

Examples

The following example shows how to configure the **load** command to indicate which phone firmware is to be used by a Cisco Unified IP Phone 7960 and 7960G, a Cisco Unified IP Phone 7912 and 7912G, and a Cisco Unified IP Phone 7941GEs. The **tftp-server** command is used to specify the location of the phone firmware files, including all firmware files for the Java-based Cisco Unified IP Phone 7941GE. Note that while no file extension is used with the **load** command, the file extension is required when using the **tftp-server** command.

```
Router(config)# voice register global
Router(config-register-global)# load 7960-7940 P00303020209
Router(config-register-global)# load 7912 P00403020209
Router(config-register-global)# load 7941 TERM41.7-0-3-0S
Router(config-register-global)# exit
Router(config)# tftp-server flash:P00303020209.bin
Router(config)# tftp-server flash:P00403020209.bin
Router(config)# tftp-server flash:SIP41.8-0-3-0S.loads
Router(config)# tftp-server flash:term61.default.loadsterm
Router(config)# tftp-server flash:41.default.loads
Router(config)# tftp-server flash:CVM41.2-0-2-26.sbn
Router(config)# tftp-server flash:cnu41.2-7-6-26.sbn
Router(config)# tftp-server flash:Jar41.2-9-2-26.sbn
```

Related Commands

Command	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.
show voice register global	Displays all global configuration parameters associated with SIP phones.
tftp-server	Enables TFTP access to firmware files on the TFTP server.
type (voice register pool)	Defines a phone type for a SIP phone.

load-cfg-file

To load configuration files on the TFTP server and to sign configuration files that are not created by Cisco Unified CME, use the **load-cfg-file** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

load-cfg-file *file-url* **alias** *file-alias* [**sign**] [**create**]
no load-cfg-file *file-url* **alias** *file-alias*

Syntax Description

file-url	Complete path of a configuration file in a local directory.
alias *file-alias*	Name of the file on the TFTP server.
sign	Signs the file and serves it on the TFTP server.
create	Creates the signed file in the local directory.

Command Default

A file is not loaded on the TFTP server.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to sign configuration files that are not created by Cisco Unified CME. This command also loads the signed and unsigned versions of the file on the TFTP server. To simply serve an already signed file on the TFTP server, use this command without the **sign** and **create** keywords.

The **create** keyword should be used with the **sign** keyword the first time that this command is used for each file. The **create** keyword is not maintained in the running configuration; this prevents signed files from being recreated during every reload.

Examples

The following example creates a file called ringlist.xml.sgn in slot0 and serves both ringlist.xml and ringlist.xml.sgn on the TFTP server.

```
telephony-service
 load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
```

The following example serves P00307010200.sbn on the TFTP server without creating a signed file.

```
telephony-service
 load-cfg-file slot0:P00307010200.sbn alias P00307010200.sbn
```

loc2

To specify the audio file used for the loss of C2 features announcement, use the **loc2** command in voice MLPP configuration mode. To disable use of this audio file, use the **no** form of this command.

loc2 *audio-url*
no loc2

Syntax Description

audio-url	Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.

Command Default

No announcement is played.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that plays to callers when the call leaves the Cisco Unified CME router on the trunk or when the user places a call to a different domain.

The **mlpp indication** command must be enabled (default) for a phone to play precedence announcements.

This command is not supported by Cisco IOS help. If you type **?**, Cisco IOS help does not display a list of valid entries.

Examples

The following example shows that the audio file played for the isolated code announcement is named ica.au located in flash:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# loc2 flash:loc2.au
```

Related Commands

Command	Description
bnea	Specifies the audio file used for the busy station not equipped for preemption announcement.
upa	Specifies the audio file used for the unauthorized precedence announcement.
vca	Specifies the audio file used for the vacant code announcement.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.

location (voice emergency response zone)

To include a location within an emergency response zone, use the **location** command in voice emergency response zone mode. To assign specific priorites to the locations, use the priority tag. To remove the location, use the **no** form of this command.

location *location-tag***[priority <1-100>]**

no location *location-tag*

Syntax Description

location-tag	Identifier for the emergency response zone location.
priority *1-100*	Identifier (1-100) for the priority ranking of locations, 1 being the highest priority.

Command Modes

Voice emergency response zone configuration (cfg-emrgncy-resp-zone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to create locations within emergency response zones. The tag must be the same as the tag that is defined using the **voice emergency response location** command. This allows routing of 911 calls to different public safety answering poins (PSAPs). Priority is optional and allows searching locations in a specified priority order. If there are locations with assigned priorities and locations configured without priorities, the prioritized locations are searched before those without an assigned priority.

Examples

The following example shows an assignment of emergency response location (ERLs) to two zones, 10 and 11, to route callers to two different PSAPs. The locations for ERLs in zone 10 are searched in sequential order for a phone address match. The calls from zone 10 have an emergency location identification number (ELIN) from ERLs 8, 9, and 10. The calls from zone 11 have an ELIN from ERLs 2, 3, 4, and 5. The locations for ERLs in zone 11 have priorities assigned and is searched in order of the assigned priority and not the ERL tag number.

```
voice emergency response zone 10
location 8
location 9
location 10
voice emergency response zone 11
location 5 priority 1
location 3 priority 2
location 4 priority 3
location 2 priority 10
```

Related Commands

Command	Description
emergency response callback	Defines a dial peer that is used for 911 callbacks from the PSAP.
emergency response location	Associates an ERL to either a SIP phone, ephone, or dial peer.
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.
voice emergency response zone	Creates an emergency response zone within which ERLs can be grouped.

log password

Effective with Cisco Unified CME 4.0, the **log password** command was replaced by the **xml user** command in telephony-service configuration mode. See the **xml user** command for more information.

For Cisco CME 3.4 and earlier versions, to set a local password for an eXtensible Markup Language (XML) Application Programming Interface (API) query, use the **log password** command in telephony-service configuration mode. To remove the password definition, use the **no** form of this command.

log password *password-string*
no log password *password-string*

Syntax Description

password-string	Character string that is a password for XML API queries. Maximum length is 28 characters. Longer strings are truncated.

Command Default

No password is defined.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was replaced by the **xml user** command.
12.4(9)T	Cisco Unified CME 4.0	This command was replaced by the **xml user** command.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Usage Guidelines

The local password is used to authenticate XML API requests on the network management server. If the password is not set, an XML API query fails local authentication.

The password string is stored as plain text. No encryption is supported.

Examples

The following example defines a local password for XML API requests:

```
Router(config)# telephony-service
Router(config-telephony)# log password ewvpil
```

log table

To set parameters for the table used to capture phone events used for the eXtensible Markup Language (XML) Application Programming Interface (API), use the **log table** command in telephony-service configuration mode. To reset parameters to their default values, use the **no** form of this command.

log table {**max-size** *entries* | **retain-timer** *minutes*}
no log table {**max-size** | **retain-timer**}

Syntax Description

max-size *entries*	Number of entries in the log table. Range is from 0 to 1000. Default is 150.
retain-timer *minutes*	Number of minutes to retain entries in the log table. Range is from 2 to 500. Default is 15.

Command Default

Default number of entries in table is 150. default number of minutes to retain entries in table is 15.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Cisco Unified CME captures and time-stamps events, such as phones registering and unregistering and extension status, and stores them in an internal buffer. This command sets the maximum number of events, or entries, that can be stored in the table. One event equals one entry. The **retain-timer** keyword sets the number of minutes that events are kept in the buffer before they are deleted.

The event table can be viewed using the **show fb-its-log** command.

Examples

The following example sets the maximum size of the table at 750 events and sets the retention time at 30 minutes:

```
Router(config)# telephony-service
Router(config-telephony)# log table max-size 750
Router(config-telephony)# log table retain-timer 30
```

Related Commands

Command	Description
show fb-its-log	Displays information about the Cisco CME XML API configuration, statistics on XML API queries, and event logs.

logging (voice emergency response settings)

To enable sylog messages to capture emergency call data, use the **logging** command in voice emergency response settings configuration mode. To disable logging, use the **no** form of this command.

logging
no logging

Syntax Description

This command has no arguments or keywords.

Command Default

This command is enabled by default.

Command Modes

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to enable syslog messages to be announced for every 911 emergency call that is made. The syslog messages can be used by third party applications to send pager or e-mail notifications to an in-house support number. This optional command is enabled by default.

Examples

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller's IP phones address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500. The outbound 911 calls do not emit a syslog message to the logging facility (for example, a local buffer, console, or remote host).

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
no logging
```

Related Commands

Command	Description
callback	Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.
elin	E.164 number used as the default ELIN if no matching ERL to the 911 caller's IP phone address is found.
expiry	Number of minutes a 911 call is associated with an ELIN in case of a callback from the 911 operator.

Command	Description
voice emergency response settings	Creates a tag for identifying settings for E911 behavior.

login (telephony-service)

To define the timer for automatically deactivating user login on SCCP phones in a Cisco Unified CME system, use the **login** command in telephony-service configuration mode. To revert to the default values for automatic logout, use the **no** form of this command.

login [**timeout** [*minutes*]] [**clear** *time*]
no login

Syntax Description

timeout	(Optional) Period of phone idleness after which user login is deactivated.
minutes	(Optional) Number of minutes for which an IP phone can be idle before the user is logged out automatically. Range: 1 to 1440. Default: 60.
clear *time*	(Optional) Time of day after which user login for all IP phones is deactivated. Range: 00:00 to 24:00 on a 24-hour clock. Default: 24:00 (midnight).

Command Default

User login is deactivated after a phone is idle for 60 minutes. User login for all phones is deactivated at 24:00.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(11)XJ2	Cisco Unified CME 4.1	Minimum value for the *minutes* argument was lowered from 5 minutes to 1 minute.
12.4(15)T	Cisco Unified CME 4.1	This command with the modifications was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command defines the after-hours login timer. Individual users on specified phones can override call blocking by logging in using a personal identification number (PIN). The after-hours login timer deactivates user login on all phones at a specific time and deactivates a login session automatically after a phone is idle for a specified period of time.

The **login** command applies only to IP phones that have soft keys, such as the Cisco Unified IP Phone 7940 and 7940G and the Cisco Unified IP Phone 7960 and 7960G.

For this command to take effect, fast reboot and reregister all phones in Cisco Unified CME by using the **restart all** command in telephony-service configuration mode.

When a Cisco Unified CME router is rebooted, the login status for all phones is reset to the default.

Examples

The following example sets the after-hours login timer to deactivate logged in phone users automatically after a 2-hour idle time and after 11:30 p.m.

```
Router(config)# telephony-service
```

```
Router(config-telephony)# login timeout 120 clear 2330
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
pin	Sets a global/individual PIN for phone users to deactivate call blocking during call blocking periods.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show ephone login	Displays the login states of all phones.

logo (voice register global)

To specify a file to display on SIP phones, use the **logo** command in voice register global configuration mode. To disable the display of the file, use the **no** form of this command.

logo *url*
no logo

Syntax Description

url	URL as defined in RFC 2396.

Command Default

No file is specified for display on idle phones.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

Use this command to define the URL for the file to be used by SIP phones connected in Cisco Unified CME. The file that is displayed must be encoded in eXtensible Markup Language (XML) by using the Cisco XML document type definition (DTD). For more information about Cisco DTD formats, see the *Cisco IP Phone Services Application Development Notes.*

After you configure this command, restart the phones by using the **reset** command.

Examples

The following example shows how to specify that the file logo.xml should be displayed on SIP phones:

```
Router(config)# voice register global
Router(config-register-global)# logo http://mycompany.com/files/logo.xml
```

Related Commands

Command	Description
reset (voice register pool)	Performs a complete reboot of one phone associated with a Cisco CME router.
reset (voice register global)	Performs a complete reboot of one or all phones associated with a Cisco CME router.

logout-profile

To enable an IP phone for extension mobility and to apply a default logout profile to the phone, use the **logout-profile** command in ephone configuration mode. To disable extension mobility, use the **no** form of this command.

logout-profile *profile-tag*
no logout-profile *profile-tag*

Syntax Description

profile-tag	Unique identifier for a default logout profile to be applied. Previously created by using the **voice logout-profile** command in voice logout-profile configuration mode. Range: 1 to maximum number of phones supported by platform.

Command Default

IP phone is not enabled for extension mobility.

Command Modes

Ephone configuration (config-ephone), Voice Register Pool configuration (voice register pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command is integrated into Cisco IOS Release 12.4(20)T.
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.

Usage Guidelines

Use this command in ephone configuration mode to enable a supported IP phone registered in Cisco Unified CME for extension mobility and to apply a default logout profile to the ephone being configured.

In Cisco Unified CME 4.2, extension mobility is supported only on SCCP IP phones.

In Cisco Unified CME 8.6 extension mobility is supported on SIP phones.

Extension mobility is not supported on non-display IP phones.

Extension mobility is not supported for analog devices.

Before using this command, you must create a logout profile to be applied to this phone by using the **voice logout-profile** command.

You cannot apply more than one logout profile to an ephone. If you attempt to apply a second logout profile to an ephone to which a profile has already been applied, the second profile will overwrite the first logout profile configuration.

Examples

The following example shows the ephone configuration for three different Cisco Unified IP phones. All three phones are enabled for extension mobility and share the same logout profile number (1), to be downloaded when these phones boot and when no phone user is logged into these phones:

```
ephone 1
 mac-address 000D.EDAB.3566
 type 7960
 logout-profile 1
ephone 2
 mac-address 0012.DA8A.C43D
 type 7970
 logout-profile 1
ephone 3
 mac-address 1200.80FC.9B01
 type 7911
 logout-profile 1
```

The following example shows the ephone configuration for two different Cisco Unified IP phones. Both phones are enabled for extension mobility and share the same logout profile number (22), to be downloaded when these phones boot and when no phone user is logged into these phones:

```
voice register pool  1
 logout-profile 22
 id mac 0012.0034.0056
 type 7960
voice register pool  2
 logout-profile 22
 id mac 0001.0023.0045
 type 7912
```

Related Commands

Command	Description
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones to which a particular logout-profile or user-profile is downloaded.
voice logout-profile	Enters voice profile configuration mode to configure a default logout profile for extension mobility.

loopback-dn

To create a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP calls and supplementary services, use the **loopback-dn** command in ephone-dn configuration mode. To delete a loopback-dn configuration, use the **no** form of this command.

loopback-dn *dn-tag* [{**forward** *number-of-digits* | **strip** *number-of-digits*}] [**prefix** *prefix-digit-string*] [**suffix** *suffix-digit-string*] [**retry** *seconds*] [**auto-con**] [**codec** {**g711alaw** | **g711ulaw**}]
no loopback-dn

Syntax Description

dn-tag	Unique sequence number that identifies the ephone-dn that is being paired for loopback with the ephone-dn that is currently being configured. The paired ephone-dn must be one that is already defined in the system.
forward *number-of-digits*	(Optional) Number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair. Range is from 1 to 32 digits. Default is to forward all digits.
strip *number-of-digits*	(Optional) Number of leading digits to be stripped from the original called number before forwarding to the other ephone-dn in the loopback-dn pair. Range is from 1 to 32 digits. Default is not to A-law strip any digits.
prefix *prefix-digit-string*	(Optional) Defines a string of digits to add in front of the forwarded called number. Maximum number of digits in the string is 32. Default is that no prefix is defined.
suffix *suffix-digit-string*	(Optional) Defines a string of digits to add to the end of the forwarded called number. Maximum number of digits in the string is 32. Default is that no suffix is defined. If you add a suffix that starts with the pound character (#), the string must be enclosed in quotation marks.
retry *seconds*	(Optional) Number of seconds to wait before retrying the loopback target when it is busy or unavailable. Range is from 0 to 32767. Default is that retry is disabled and appropriate call-progress tones are passed to the call originator.
auto-con	(Optional) Immediately connects the call and provides in-band alerting while waiting for the far-end destination to answer. Default is that automatic connection is disabled.
codec	(Optional) Explicitly forces the G.711 A-law or G.711 mu-law voice coding type to be used for calls that pass through the loopback-dn. This overrides the G.711 coding type that is negotiated for the call and provides mu-law to A-law conversion if needed. Default is that Real-Time Transport Protocol (RTP) voice packets are passed through the loopback-dn without considering the G.711 coding type negotiated for the calls.
g711alaw	G.711 A-law, 64000 bits per second, for T1.
g711ulaw	G.711 mu-law, 64000 bits per second, for E1.

Command Default

All calls are set to forward all digits and not to strip any digits. Prefix is not defined. Suffix is not defined. Retry is disabled. Automatic connection is disabled. RTP voice packets are passed through the loopback-dn without considering the G.711 coding type negotiated for the call.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(2)XT3	Cisco ITS 2.0	The **suffix** keyword was added.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and the **auto-con** keyword was added.
12.2(11)T	Cisco ITS 2.01	The **suffix** keyword was added**.**
12.2(11)YT	Cisco ITS 2.1	The **strip** keyword was added.
12.2(11)YT2	Cisco ITS 2.1	The **codec** keyword was added.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

The **loopback-dn** command is used to configure two ephone-dn virtual voice ports as back-to-back-connected voice-port pairs. A call presented on one side of the loopback-dn pair is reoriginated as a new call on the opposite side of the loopback-dn pair. The **forward**, **strip, prefix**, and **suffix** keywords can be used to manipulate the original called number that is presented to the incoming side of the loopback-dn pair to generate a modified called number to use when reoriginating the call at the opposite side of the loopback-dn pair. For loopback-dn configurations, you must always configure ephone-dn virtual voice ports as cross-coupled pairs.

Note

Use of loopback-dn configurations within a VoIP network should be restricted to resolving critical network interoperability service problems that cannot otherwise be solved. Loopback-dn configurations are intended to be used in VoIP network interworking situations in which the only other alternative would be to make use of back-to-back-connected physical voice ports. Loopback-dn configurations emulate the effect of a back-to-back physical voice-port arrangement without the expense of the physical voice-port hardware. A disadvantage of loopback-dn configurations is that, because digital signal processors (DSPs) are not involved in a loopback-dn arrangement, the configuration does not support interworking or transcoding between calls that use different voice codecs. In many cases, the use of back-to-back physical voice ports that do use DSPs to resolve VoIP network interworking issues is preferred, because it introduces fewer restrictions in terms of supported codecs and call flows. Also, loopback-dns do not support T.38 fax relay.

Note

We recommend that you create the basic ephone-dn configuration for both ephone-dn entries before configuring the loopback-dn option under each ephone-dn. The loopback-dn mechanism should be used only in situations where the voice call parameters for the calls on either side of the loopback-dn use compatible configurations; for example, compatible voice codec and dual tone multifrequency (DTMF) relay parameters. Loopback-dn configurations should be used only for G.711 voice calls.

The loopback-dn arrangement allows an incoming telephone call to be terminated on one side of the loopback-dn port pair and a new pass-through outgoing call to be originated on the other side of the loopback-dn port pair. The loopback-dn port pair normally works with direct cross-coupling of their call states; the alerting call state on the outbound call segment is associated with the ringing state on the inbound call segment.

The loopback-dn mechanism allows for call operations (such as call transfer and call forward) that are invoked for the call segment on one side of the loopback-dn port pair to be isolated from the call segment that is present on the opposite side of the loopback-dn port pair. This approach is useful when the endpoint devices associated with the two different sides have mismatched call-transfer and call-forwarding capabilities. The loopback-dn arrangement allows for call-transfer and call-forward requests to be serviced on one side of the loopback-dn port pair by creating hairpin-routed calls when necessary. The loopback-dn arrangement avoids the propagation of call-transfer and call-forward requests to endpoint devices that do not support these functions.

The **loopback-dn** command provides options for controlling the called-number digits that are passed through from the incoming side to the outgoing side. The available digits can be manipulated with the **forward**, **strip**, **prefix**, and **suffix** keywords.

The **forward** keyword defines the number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair. The default is set to forward all digits. The **strip** keyword defines the number of leading digits to be stripped from the original called number before forwarding to the other ephone-dn in the loopback-dn pair. The default is set to not strip any digits. The **forward** and **strip** commands are mutually exclusive and can be used with any combination of the **prefix** and **suffix** keywords.

The **prefix** keyword defines a string of digits to add in front of the forwarded number.

The **suffix** keyword is most commonly used to add a terminating "#" (pound-sign) character to the end of the forwarded number to indicate that no more digits should be expected. The pound-sign character indicates to the call-routing mechanism that is processing the forwarded number that the forwarded number is complete. Providing an explicit end-of-number character also avoids a situation in which the call-processing mechanism waits for the interdigit timeout period to expire before routing the call onward using the forwarded number.

Note The Cisco IOS command-line interface (CLI) requires that arguments with character strings that start with the pound-sign (#) character be enclosed within quotation marks; for example, "#".

The **retry** keyword is used to suppress a far-end busy indication on the outbound call segment. Instead of returning a busy signal to the call originator (on the incoming call segment), a loopback-dn presents an alerting or ringing tone to the caller and then periodically retries the call to the final far-end destination (on the outgoing call segment). This is not bidirectional. To prevent calls from being routed into the idle outgoing side of the loopback-dn port pair during the idle interval that occurs between successive outgoing call attempts, configure the outgoing side of the loopback-dn without a number so that there is no number to match for the inbound call.

The **auto-con** keyword is used to configure a premature trigger for a connected state for an incoming call segment while the outgoing call segment is still in the alerting state. This setup forces the voice path to open for the incoming call segment and support the generation of in-band call progress tones for busy, alerting, or ringback. The disadvantage of the **auto-con** keyword is premature opening of the voice path during the alerting stage and also triggering of the beginning of billing for the call before the call has been answered by the far end. These disadvantages should be considered carefully before you use the **auto-con** keyword.

The **codec** keyword is used to explicitly select the A-law or mu-law type of G.711 and to provide A-law to mu-law conversion if needed. Setting the codec type on one side of the loopback-dn forces the selection of A-law or mu-law for voice packets that are transmitted from that side of the loopback-dn. To force the A-law or mu-law G.711 codec type for both voice packet directions, set the codec type on both sides of the loopback-dn. Loopback-dn configurations are used only with G.711 calls. Other voice codec types are not supported.

Examples

The following example creates a loopback-dn configured with the **forward** and **prefix** keywords:

```
Router(config)# ephone-dn 7
Router(config-ephone-dn)# loopback-dn 15 forward 5 prefix 41
```

The following example creates a loopback-dn that appends the pound-sign (#) character to forwarded numbers to indicate the end of the numbers:

```
Router(config)# ephone-dn 7
Router(config-ephone-dn)# loopback-dn 16 suffix "#"
```

The following example shows a loopback-dn configuration that pairs ephone-dns 15 and 16. An incoming call (for example, from VoIP) to 4085550101 matches ephone-dn 16. The call is then reoriginated from ephone-dn 15 and sent to extension 50101. Another incoming call (for example, from a local IP phone) to extension 50151 matches ephone-dn 15. It is reoriginated from ephone-dn 16 and sent to 4085550151.

```
ephone-dn 15
 number 5015.
 loopback-dn 16 forward 5 prefix 40855
 caller-id local
 no huntstop
!
ephone-dn 16
 number 408555010.
 loopback-dn 15 forward 5
 caller-id local
 no huntstop
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
show ephone-dn loopback	Displays information about loopback ephone-dns that have been created in a Cisco CME system.

lpcor incoming

To associate an incoming call with a logical partitioning class of restriction (LPCOR) resource-group policy, use the **lpcor incoming** command in ephone, ephone-template, trunk group, voice port, voice register pool, voice register template, or voice service configuration mode. To reset to the default, use the **no** form of this command.

lpcor incoming *lpcor-group*
no lpcor incoming

Syntax Description

lpcor-group	Name of the LPCOR resource group.

Command Default

LPCOR policy is not associated with the incoming call.

Command Modes

Ephone configuration (config-ephone)
Ephone template configuration (config-ephone-template)
Trunk group configuration (config-trunk-group)
Voice port configuration (config-voiceport)
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)
Voice service configuration (conf-voi-serv)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Analog Phones

An incoming call to an analog phone is associated with the LPCOR policy specified with this command in the voice port. Otherwise the LPCOR policy specified in the trunk group is used.

SCCP IP Phones (Local or Remote)

An incoming call to an SCCP IP phone is associated with the LPCOR policy specified with this command in ephone or ephone template configuration mode. The ephone configuration has precedence over the ephone-template configuration. All directory numbers on the phone share the same LPCOR setting.

SIP IP Phones (Local or Remote)

An incoming call to a SIP IP phone is associated with the LPCOR policy specified with this command in voice register pool or voice register template configuration mode. The voice register pool configuration has precedence over the voice register template configuration. All directory numbers on the phone share the same LPCOR setting.

Note This command is not supported for phones configured with the **lpcor type mobility** command.

- Phones that share a directory number must be configured with the same LPCOR policy. Different LPCOR settings on shared-line phones are not supported.

PSTN Trunks

An incoming call to the PSTN is associated with the LPCOR policy specified with this command in the voice port. Otherwise the LPCOR policy specified in the trunk group is used. The voice port configuration takes precedence.

VoIP Trunks (H.323 or SIP)

An incoming call to a VoIP trunk is associated with the LPCOR policy specified with this command in voice service configuration mode if the remote IP address is not found in the IP trunk subnet table created with the **voice lpcor ip-trunk subnet incoming** command.

Examples

The following example shows the command used in different configuration modes:

```
voice service voip
 lpcor incoming voip_group1
!
trunk group analog1
 lpcor incoming analog_group1
 lpcor outgoing analog_group1
!
voice-port 1/1/0
 lpcor incoming vp_group1
 lpcor outgoing vp_group1
!
voice register pool  3
 lpcor type remote
 lpcor incoming sip_group3
 lpcor outgoing sip_group3
 id mac 001E.BE8F.96C0
 type 7940
 number 1 dn 3
!
ephone 2
 mac-address 001C.821C.ED23
 type 7960
 button 1:2
 lpcor type remote
 lpcor incoming ephone_group2
 lpcor outgoing ephone_group2
```

Related Commands

Command	Description
lpcor outgoing	Associates an outgoing call with a LPCOR resource-group policy.
lpcor type	Specifies the LPCOR type for an IP phone.
voice lpcor ip-trunk subnet incoming	Creates a LPCOR IP-trunk subnet table for incoming calls from a VoIP trunk.
voice lpcor policy	Creates a LPCOR policy for a resource group.

lpcor outgoing

To associate an outgoing call with a logical partitioning class of restriction (LPCOR) resource-group policy, use the **lpcor outgoing** command in dial peer, ephone, ephone template, trunk group, voice port, voice register pool, or voice register template configuration mode. To reset to the default, use the **no** form of this command.

lpcor outgoing *lpcor-group*
no lpcor outgoing

Syntax Description

lpcor-group	Name of the LPCOR resource group.

Command Default

LPCOR policy is not associated with the outgoing call.

Command Modes

Dial peer configuration (config-dial-peer)
Ephone configuration (config-ephone)
Ephone template configuration (config-ephone-template)
Trunk group configuration (config-trunk-group)
Voice port configuration (config-voiceport)
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Analog Phones

An outgoing call from an analog phone is associated with the LPCOR policy specified with this command in the voice port. Otherwise the LPCOR policy specified in the trunk group is used.

SCCP IP Phones (Local or Remote)

An outgoing call from an SCCP IP phone is associated with the LPCOR policy specified with this command in ephone configuration or ephone template configuration mode. The ephone configuration has precedence over the ephone-template configuration. All directory numbers on the phone share the same LPCOR setting.

SIP IP Phones (Local or Remote)

An outgoing call from a SIP IP phone is associated with the LPCOR policy specified with this command in voice register pool or voice register template configuration mode. The voice register pool configuration has precedence over the voice register template configuration. All directory numbers on the phone share the same LPCOR setting.

Note This command is not supported for phones configured with the **lpcor type mobility** command.

- Phones that share a directory number must be configured with the same LPCOR policy. Different LPCOR settings on shared-line phones are not supported.

PSTN Trunks

An outgoing call from the PSTN uses the LPCOR policy specified with this command in the voice port if the outbound dial peer is configured with the **port** command. Otherwise the outgoing call uses the LPCOR policy specified with this command in the trunk group if the outbound dial peer is configured with the **trunkgroup** command.

VoIP Trunks (H.323 or SIP)

An outgoing VoIP call uses the LPCOR policy specified with this command in the outbound dial peer. Otherwise the outgoing call uses the default LPCOR policy.

Examples

The following example shows the command used in different configuration modes:

```
trunk group analog1
 lpcor incoming analog_group1
 lpcor outgoing analog_group1
!
voice-port 1/1/0
 lpcor incoming vp_group1
 lpcor outgoing vp_group1
!
dial-peer voice 2 voip
 destination-pattern 2...
 lpcor outgoing voip_group2
 session protocol sipv2
 session target ipv4:192.168.97.1
!
voice register pool  3
 lpcor type remote
 lpcor incoming sip_group3
 lpcor outgoing sip_group3
 id mac 001E.BE8F.96C0
 type 7940
 number 1 dn 3
!
ephone 2
 mac-address 001C.821C.ED23
 type 7960
 button 1:2
 lpcor type remote
 lpcor incoming ephone_group2
 lpcor outgoing ephone_group2
```

Related Commands

Command	Description
lpcor incoming	Associates an incoming call with a LPCOR resource-group policy.
lpcor type	Specifies the LPCOR type for an IP phone.
port (dial-peer)	Associates a dial peer with a voice port.
trunkgroup	Associates a dial peer with a trunk group.
voice lpcor policy	Creates a LPCOR policy for a resource group.

lpcor type

To specify the logical partitioning class of restriction (LPCOR) type for an IP phone, use the **lpcor type** command in ephone, ephone-template, voice register pool, or voice register template configuration mode. To reset to the default, use the **no** form of this command.

lpcor type {**local** | **mobile** | **remote**}
no lpcor type

Syntax Description

local	IP phone always registers to Cisco Unified CME through the LAN.
mobile	IP phone can register to Cisco Unified CME through the LAN or WAN.
remote	IP phone always registers to Cisco Unified CME through the WAN.

Command Default

LPCOR feature is disabled for the IP phone.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

According to the Telecom Regulatory Authority of India (TRAI) requirements, an IP phone can accept both PSTN and VoIP calls if it is locally registered to Cisco Unified CME through the LAN. Select the **local** keyword for this type of phone.

If an IP phone is registered remotely to Cisco Unified CME through the WAN, PSTN calls must be blocked from that remote IP phone. Select the **remote** keyword for this type of phone.

A static LPCOR policy is applied to an IP phone if the phone registers to Cisco Unified CME from the same region (local or remote) permanently.

If an IP phone moves between the local and remote regions, such as an Extension Mobility phone, Cisco IP Communicator softphone, or remote teleworker, select the **mobile** keyword. The LPCOR policy is assigned dynamically based on the phone's currently registered IP address.

If you use a phone template to apply a command to a phone and you also use the same command in the phone configuration of the same phone, the value in phone configuration has priority.

Examples

The following example shows that SCCP IP phone 2 is set to the remote LPCOR type:

```
ephone 2
 mac-address 001C.821C.ED23
 type 7960
 button 1:2
```

```
lpcor type remote
lpcor incoming ephone_group2
lpcor outgoing ephone_group2
```

Related Commands

Command	Description
lpcor incoming	Associates a LPCOR resource-group policy with an incoming call.
lpcor outgoing	Associates a LPCOR resource-group policy with an outgoing call.
voice lpcor policy	Creates a LPCOR policy for a resource group.

Cisco Unified CME Commands: M

mac-address (ephone)

To associate the MAC address of a Cisco IP phone with an ephone configuration in a Cisco CallManager Express (Cisco CME) system, use the **mac-address** command in ephone configuration mode. To disassociate the MAC address from an ephone configuration, use the **no** form of this command.

mac-address [*mac-address*] [**reserved**]
no mac-address

Syntax Description

mac-address	Identifying MAC address of an IP phone, which is found on a sticker located on the bottom of the phone.
reserved	Identifies the reserved MAC address of the phone.

Command Default

There are no default behavior or values for this command.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)ZJ	Cisco CME 3.0	The *mac-address* argument was made optional to enable automatic MAC address assignment after registration of phones.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Use this command to specify the MAC address of a specific Cisco IP phone in order to physically identify the Cisco IP phone in a Cisco CME configuration. The MAC address of each Cisco IP phone is printed on a sticker that is placed on the bottom of the phone.

If you choose to register phones before configuring them, the **mac-address** command can be used during configuration without entering the *mac-address* argument. The Cisco CME system detects MAC addresses and automatically populates phone configurations with their corresponding MAC addresses and phone types. This capability is not supported for voice-mail ports and is supported only by Cisco CME 3.0 and later versions. To use this capability, enable Cisco CME by using the following commands: **max-ephones**, **max-dn**, **create cnf-files**, and **ip source-address.** After these commands have been used, phones can start to register. Then, when you are configuring a registered ephone and you use the **mac-address** command with no argument, the MAC address of the phone is automatically read into the configuration. The equivalent functionality is available through the Cisco CME graphic user interface (GUI).

If you choose to configure phones before registering them, the MAC address for each ephone must be entered during configuration.

Examples

The following example associates the MAC address CFBA.321B.96FA with the IP phone that has phone-tag 22:

```
Router(config)# ephone 22
Router(config-ephone)# mac-address CFBA.321B.96FA
```

Related Commands

	Description
create cnf-files	Builds the XML configuration files that are required for IP phones used with Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or later versions.
ip source-address	Identifies the IP address and port through which IP phones communicate with a Cisco CME router.
max-dn	Sets the maximum number of ephone-dns to be supported by a Cisco CME router.
max-ephones	Sets the maximum number of ephones to be supported by a Cisco CME router.
show ephone registered	Displays status and information for registered IP phones.

mac-address (voice-gateway)

To define the MAC address of the voice gateway to autoconfigure, use the **mac-address** command in voice-gateway configuration mode. To remove the MAC address from the configuration, use the **no** form of this command.

mac-address *mac-address*
no mac-address

Syntax Description

mac-address	MAC address of the voice gateway.

Command Default

No MAC address is defined for the voice gateway to be autoconfigured.

Command Modes

Voice-gateway configuration (config-voice-gateway)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS release 12.4(24)T.

Usage Guidelines

This command defines the MAC address of the Cisco voice gateway that downloads its XML configuration file from Cisco Unified CME using the Autoconfiguration feature.

Examples

The following example associates the MAC address 001F.A30F.8331 for the Cisco VG224 voice gateway associated with tag 1:

```
voice-gateway system  1
 network-locale FR
 type VG224
 mac-address 001F.A30F.8331
 voice-port 0-23
 create cnf-files
```

Related Commands

Command	Description
type (voice-gateway)	Defines the type of voice gateway to autoconfigure in Cisco Unified CME.
voice-port (voice-gateway)	Identifies the analog ports on the voice gateway that register to Cisco Unified CME.

mailbox-selection (dial-peer)

To set a policy for selecting a mailbox for calls from a Cisco Unified CME system that are diverted before being sent to a Cisco Unity Express or PBX voice-mail pilot number, use the **mailbox-selection** command in dial-peer configuration mode. To return to the default, use the **no** form of this command.

mailbox-selection {**last-redirect-num** | **orig-called-num**}
no mailbox-selection

Syntax Description

last-redirect-num	(PBX voice mail only) The mailbox to which the call will be sent is the number that diverted the call to the voice-mail pilot number (the last number to divert the call).
orig-called-num	(Cisco Unity Express only) The mailbox to which the call will be sent is the number that was originally dialed before the call was diverted.

Command Default

Cisco Unity Express uses the last number to which the call was diverted before it was sent to voice mail as the mailbox number. Some legacy PBX systems use the originally called number as the mailbox number.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

When Cisco Unified CME diverts a call, it captures the reroute information which will be used to compose a reroute request. A dial-peer match will be performed against the diverted-to number. If this is the voice mail pilot number and the **mailbox-selection** command has been used to install a policy, the reroute information will be amended as directed by the command. The originator will pick up the modified reroute request, build the diversion information and include it in the new diverted call to the voice-mail pilot number.

This command should be used on the outbound dial peer for the pilot number of the voice-mail system.

This command might not work properly in certain network topologies, including the following cases:

- When the last redirecting endpoint is not hosted on Cisco Unified CME. This rarely occurs with a PBX.
- When a call is forwarded across several SIP trunks. Multiple SIP Diversion Headers (stacking hierarchy) are not supported in Cisco IOS software.
- When a call is forwarded across non Cisco voice gateways that do not support the optional H450.3 originalCalledNr field.

Examples

The following example shows how to set a policy to select the mailbox of the originally called number when a call is diverted to a Cisco Unity Express or PBX voice-mail system with the pilot number 7000.

```
dial-peer voice 7000 voip
 destination-pattern 7000
 session target ipv4:10.3.34.211
 codec g711ulaw
```

```
no vad
mailbox-selection orig-called-num
```

mailbox-selection (ephone-dn)

To set a policy for selecting a mailbox for calls that are diverted before being sent to a Cisco Unity voice-mail pilot number, use the **mailbox-selection** command in ephone-dn configuration mode. To return to the default, use the **no** form of this command.

mailbox-selection last-redirect-num
no mailbox-selection

Syntax Description

last-redirect-num	The mailbox to which the call will be sent is the last number to divert the call.

Command Default

Cisco Unity uses the originally called number as the mailbox number.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command sets the policy for selecting a mailbox for diverted calls.

This command is used on the ephone-dn associated with the voice-mail pilot number.

This command can only be used with SCCP phones.

This command might not work properly in certain network topologies, including the following cases:

- When the last redirecting endpoint is not hosted on Cisco Unified CME. This may rarely occur with a PBX.
- When a call is forwarded across several SIP trunks. Multiple SIP Diversion Headers (stacking hierarchy) are not supported in Cisco IOS software.
- When a call is forwarded across non Cisco voice gateways that do not support the optional H450.3 originalCalledNr field.

Examples

The following example sets a policy to select the mailbox of the last redirecting number when a call is diverted to a Cisco Unity voice-mail system with the pilot number 8000.

```
ephone-dn 2583
 number 8000
 mailbox-selection last-redirect-num
```

max-calls-per-button

To set the maximum number of calls allowed on an octo-line directory number on an SCCP phone, use the **max-calls-per-button** command in ephone or ephone-template configuration mode. To reset to the default, use the **no** form of this command.

max-calls-per-button *number-of-calls*
no max-calls-per-button

Syntax Description

number-of-calls	Maximum number of calls. Range: 1 to 8. Default: 8.

Command Default

Maximum number of calls allowed on an octo-line is 8.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 4.3	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command limits the maximum number of calls, both incoming and outgoing, that can be active on each octo-line directory number on an SCCP phone. This command applies to all octo-line directory numbers on the phone.

This command must be set to a value that is more than or equal to the value set with the **busy-trigger-per-button** command.

For phones that do not support octo-line directory numbers such as the Cisco Unified IP Phone 7902, 7920, or 7931, and analog phones connected to the Cisco VG224 or Cisco ATA, we recommend that you set the **max-calls-per-button** command to 2. Otherwise, after the phone type is identified with either the **type** command or during phone registration, this command is automatically set to 2.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode for the same ephone, the value that you set in ephone configuration mode has priority.

Examples

The following example sets the maximum calls allowed on octo-lines to 4 on ephone 1.

```
Router(config)#
ephone 1
Router(config-ephone)# max-calls-per-button 4
```

Related Commands

Command	Description
busy-trigger-per-button	Sets the maximum number of incoming calls allowed on an octo-line directory number before it triggers Call Forward Busy on the phone.

Command	Description
ephone-dn	Configures a directory number for SCCP phones.
type	Assigns a phone type to an SCCP phone.

max-conferences

To set the maximum number of three-party conferences that are supported simultaneously by the Cisco CallManager Express (Cisco CME) router, use the **max-conferences** command in telephony-service configuration mode. To reset this number to the default, use the **no** form of this command.

max-conferences*max-conference-number***[gain -6 | 0 | 3 | 6]**
no max-conferences

Syntax Description

max-conference number	Maximum number of three-party conferences that are supported simultaneously by the router. This number is platform-dependent, and the default is half the maximum for each platform. The following are the maximum values for this argument: • Cisco 1700 series, Cisco 2600 series, Cisco 2801—8 • Cisco 2811, Cisco 2821, Cisco 2851, Cisco 3600 series, Cisco 3700 series—16 • Cisco 3800 series—24 (requires Cisco IOS Release 12.3(11)XL or higher) **Note** Each individual Cisco IP phone can host a maximum of one conference at a time. You cannot create a second conference on the phone if you already have an existing conference on hold.
gain	(Optional) Increases the sound volume of VoIP and public switched telephony network (PSTN) parties joining a conference call. The allowable decibel units are -6 db, 0 db, 3 db, and 6 db. The default is -6 db.

Command Default

Default is half the maximum number of simultaneous three-party conferences for each platform.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.3(11)XL1	Cisco CME 3.2.1	The **gain** keyword was added.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command supports three-party conferences for local and on-net calls only when all conference participants are using the G.711 codec. Conversion between G.711 mu-law and A-law is supported. Mixing of the media streams is supported by the Cisco IOS processor. The maximum number of simultaneous conferences is limited to the platform-specific maximums.

The **gain** keyword's functionality is applied to inbound audio packets, so conference participants can more clearly hear a remote PSTN or VoIP caller joining their call. Note that this functionality cannot discriminate between a remote VoIP/foreign exchange office (FXO) source, which requires a volume gain, and a remote VoIP/IP phone, which does not require a volume gain and may therefore incur some sound distortions.

Examples

The following example sets the maximum number of conferences for a Cisco IP phone to 4 and configures a gain of 6 db for inbound audio packets from remote PSTN or VoIP calls joining a conference:

```
Router(config)# telephony-service
Router(config-telephony)# max-conferences 4 gain 6
```

max-dn

To set the maximum quantity of directory numbers (ephone-dns) that can be configured on a Cisco Unified CME router, use the **max-dn** command in telephony-service configuration mode. To reset this number to the default value, use the **no** form of this command.

max-dn *max-directory-numbers* [**preference** *preference-order*] [**no-reg** {**primary** | **both**}]
no max-dn

Syntax Description

maxdirectorynumbers	Maximum number of directory numbers (ephone-dns) to allow in the Cisco CME system. The maximum you can set depends on the software version, router platform, and amount of memory that you have installed. Type **?** to display range. The default is 0.
preference *preference-order*	(Optional) Sets a preference value for the primary number of an ephone-dn. Refer to CLI help for a range of numeric options, where 0 is the highest preference. Default is 0.
no-reg	(Optional) Globally disables ephone registration with an H.323 gatekeeper or SIP proxy.
primary	Primary ephone-dn numbers only.
both	Both primary and secondary ephone-dn numbers.

Command Default

The default is 0.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	Cisco Unified 4.0	The **preference**, **no-reg**, **primary**, and **both** keywords were introduced.
12.4(9)T	Cisco Unified 4.0	The **preference**, **no-reg**, **primary**, and **both** keywords were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **max-dn** command limits the number of extensions (ephone-dns) available in a Cisco Unified CME system. The maximum number of ephone-dns that you can create depends on the software version, router platform, and amount of memory that you have installed. Type **?** to display range.

The **max-ephones** command similarly limits the number of IP phones in a Cisco Unified CME system.

> **Note**
> You can increase the number of allowable extensions to the maximum; but after the maximum allowable number is configured, you cannot reduce the limit without rebooting the router.

If registration with an H.323 gatekeeper or SIP proxy is enabled globally (the default), you can override the setting per extension by using the **no-reg** keyword in the **number** command for individual ephone-dns.

After using this command, you can provision individual extensions using the Cisco Unified CME graphic user interface (GUI) or the router CLI in ephone-dn configuration mode.

Examples

The following example sets the maximum number of extensions (ephone-dns) to 12:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 12
```

The following example sets the maximum number of extensions to 150 and specifies that the primary number of each extension should receive a dial-peer preference order of 1:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 150 preference 1
```

The following example sets the maximum number of extensions to 200 and specifies that they should not register both primary and secondary numbers with the H.323 gatekeeper:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 200 no-reg both
```

The following example sets the maximum number of extensions to 200 and specifies that ephone-dn 36 should not register its primary number with the gatekeeper:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 200
Router(config-telephony)# exit
Router(config)# ephone-dn 36
Router(config-ephone-dn)# number 75373 no-reg primary
```

Related Commands

	Description
ephone-dn	Enters ephone-dn configuration mode.
max-ephones	Sets the maximum number of phones supported by the router.
number	Associates a telephone or extension number with an ephone-dn.

max-dn (voice register global)

To set the maximum number of SIP phone directory numbers (extensions) that are supported by a Cisco router, use the **max-dn** command in voice register global configuration mode. To reset to the default, use the **no** form of this command.

max-dn *max-directory-numbers*
no max-dn

Syntax Description

maxdirectorynumbers	Maximum number of extensions (ephone-dns) supported by the Cisco router. The maximum number is version and platform dependent; type **?** to display range. • In Cisco CME 3.4 to Cisco Unified CME 7.0 and in Cisco SIP SRST 3.4 to Cisco Unified SIP SRST 7.0: Default is maximum number supported by platform. • In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions: Default is 0.

Command Default

Before Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1, default is maximum number supported by platform.

In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions, default is 0.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(22)YB	Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1	The default value was changed to 0.
12.4(24)T	Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1	This command was integrated into Cisco IOS release 12.4(24)T.

Usage Guidelines

This command limits the number of SIP phone directory numbers (extensions) available in a Cisco Unified CME system. The **max-dn** command is platform specific. It defines the limit for the **voice register dn** command. The **max-pool** command similarly limits the number of SIP phones in a Cisco CME system.

You can increase the number of allowable extensions to the maximum; but after the maximum allowable number is configured, you cannot reduce the limit without rebooting the router. You cannot reduce the number of allowable extensions without removing the already-configured directory numbers with dn-tags that have a higher number than the maximum number to be configured.

Note This command can also be used for Cisco Unified SIP SRST.

Examples

The following example shows how to set the maximum number of directory numbers to 48:

```
Router(config)# voice register global
Router(config-register-global)# max-dn 48
```

Related Commands

Command	Description
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
max-pool (voice register global)	Sets the maximum number of SIP voice register pools that are supported in a Cisco SIP SRST or Cisco CME environment.

max-ephones

To set the maximum number of Cisco IP phones to be supported by a Cisco CallManager Express (Cisco CME) router, use the **max-ephones** command in telephony-service configuration mode. To reset this number to the default value, use the **no** form of this command.

max-ephones *max-phones*
no max-ephones

Syntax Description

maxphones	Maximum number of phones supported by the Cisco CME router. The maximum number is version- and platform-dependent; refer to Cisco IOS command-line interface (CLI) help. Default is 0.

Command Default

Default is 0.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(15)XZ	Cisco Unified CME 4.3	This command was modified to set the maximum number of phones that can register to Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

The **max-ephones** command limits the number of Cisco IP phones supported on the router. The maximum number you can set is platform- and version-dependent. Use CLI help to determine the maximum number of ephones you can set, as shown in this example:

```
Router(config-telephony)# max-ephones ?
  <1-48>  Maximum phones to support
```

The **max-dn** command similarly limits the number of extensions (ephone-dns) in a Cisco CME system.

Note You can increase the number of phones; but after the maximum allowable number is configured, you cannot reduce the limit of the Cisco IP phones without rebooting the router.

After using this command, configure phones by using the Cisco CME graphic user interface (GUI) or the router CLI in ephone configuration mode.

Examples

The following example sets the maximum number of Cisco IP phones in a Cisco CME system to 24:

```
Router(config)# telephony-service
Router(config-telephony)# max-ephones 24
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.
max-dn	Sets the maximum number of extensions (ephone-dns) that can be supported by the router.

max-idle-time

To create an idle-duration timer for automatically logging out an Extension Mobility user, use the **max-idle-time** command in voice user-profile configuration mode. To remove the timer, use the **no** form of this command.

max-idle-time *minutes*
no max-idle-time

Syntax Description

minutes	Maximum number of minutes an Extension Mobility phone is idle after which the logged-in user is logged out from Extension Mobility. Range:1 to 9999.

Command Default

No timer is created.

Command Modes

Voice user-profile configuration (config-user-profile)

Command History

Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command creates an idle-duration timer for automatically logging a user out from Extension Mobility. The timer monitors the phone and if the specified maximum idle time is exceeded, the EM manager logs out the user. Typically this command is used to log out users who fail to manually log out of Extension Mobility before leaving a phone.

The call history record is automatically cleared when a user logs out from an Extension Mobility phone. To disable Automatic Clear Call History on all Extension Mobility phones, use the **keep call-history** command in telephony-service configuration mode.

After creating or modifying a profile, use the **reset** command in voice user-profile configuration mode to reset all phones on which this profile is downloaded to propagate the modifications.

Examples

The following example shows how to create a 30-minute idle-duration timer in user profile 1:

```
Router(config)# voice user-profile 1
Router(config-user-profile)# max-idle-time 30
Router(config-user-profile)# reset
Router(config-user-profile)#
```

Related Commands

Command	Description
keep call-history	Disables Automatic Clear Call History for Extension Mobility in Cisco Unified CME.
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones on which a particular logout profile or user profile is downloaded.

maximum bit-rate (video)

To modify the maximum IP phone video bandwidth in Cisco Unified CME, use the **maximum bit-rate** command in video configuration mode. To restore the default maximum bit-rate, use the **no** form of this command.

maximum bit-rate *value*
no maximum bit-rate

Syntax Description

value	Video bandwidth in kb/s Range is 0 to 10000000. Default value is 10000000.

Command Default

Maximum bit-rate of video bandwidth is 1,000,000 kb/s.

Command Modes

Video configuration (config-tele-video)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to modify the default value of the maximum video bandwidth for all video-capable phones associated with a Cisco Unified CME router. Default value is 1,000,000 kb/s.

Examples

The following example sets a maximum bit-rate of 256 kb/s.

```
Router(config)#
telephony-service
Router(config-telephony)# video
Router(conf-tele-video)# maximum bit-rate 256
```

max-pool (voice register global)

To set the maximum number of Session Initiation Protocol (SIP) voice register pools that are supported in Cisco Unified SIP SRST or Cisco Unified CME, use the **max-pool** command in voice register global configuration mode. To reset the maximum number to the default, use the **no** form of this command.

max-pool *max-voice-register-pools*
no max-pool

Syntax Description

maxvoice-register-pools	Maximum number of SIP voice register pools supported by the Cisco router. The upper limit of voice register pools is version- and platform-dependent; type **?** for range. • In Cisco CME 3.4 to Cisco Unified CME 7.0 and in Cisco SIP SRST 3.4 to Cisco Unified SIP SRST 7.0: Default is maximum number supported by platform. • In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions: Default is 0.

Command Default

Before Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1, default is maximum number supported by platform. In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions, default is 0.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(22)YB	Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1	The default value was changed to 0.
12.4(24)T	Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1	This command was integrated into Cisco IOS release 12.4(24)T.

Usage Guidelines

This command limits the number of SIP phones supported by Cisco Unified CME. The **max-pool** command is platform specific and defines the limit for the **voice register pool** command.

The **max-dn** command similarly limits the number of directory numbers (extensions) in Cisco Unified CME.

You can increase the number of phones; but after the maximum allowable number is configured, you cannot reduce the limit of the SIP phones without rebooting the router.

Note This command can also be used for Cisco Unified SIP SRST.

Examples

The following example shows how to set the maximum number of Cisco SIP IP phones in Cisco Unified SIP SRST or Cisco Unified CME to 24:

```
Router(config)# voice register global
Router(config-register-global)# max-pool 24
```

Related Commands

Command	Description
max-dn (voice register global)	Set the maximum number of SIP phone directory numbers (extensions) that are supported by a Cisco Unified CME router.

max-presentation

To set the number of call presentation lines supported by a phone type, use the **max-presentation** command in ephone-type configuration mode. To reset to the default, use the **no** form of this command.

max-presentation *number*
no max-presentation

Syntax Description

number	Number of presentation lines. Range: 1 to 100. Default: 0. See the table for the number of presentation lines supported by each phone type.

Command Default

No display lines are supported by the phone type.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command defines the number of presentation lines that are supported for the type of phone being added with an ephone-type template.

Table 11: Supported Values for Ephone-Type Commands

Supported Device	device-id	device-type	num-buttons	max-presentation
Cisco Unified IP Phone 6901	547	6901	1	1
Cisco Unified IP Phone 6911	548	6911	1	10
Cisco Unified IP Phone 7915 Expansion Module with 12 buttons	227	7915	12	0 (default)
Cisco Unified IP Phone 7915 Expansion Module with 24 buttons	228	7915	24	0
Cisco Unified IP Phone 7916 Expansion Module with 12 buttons	229	7916	12	0
Cisco Unified IP Phone 7916 Expansion Module with 24 buttons	230	7916	24	0
Cisco Unified Wireless IP Phone 7925	484	7925	6	4
Cisco Unified IP Conference Station 7937G	431	7937	1	6
Nokia E61	376	E61	1	1

Examples

The following example shows that 1 presentation line is specified for the Nokia E61 when creating the ephone-type template.

```
Router(config)# ephone-type E61
Router(config-ephone-type)# max-presentation 1
```

Related Commands

Command	Description
device-id	Specifies the device ID for a phone type in an ephone-type template.
num-buttons	Sets the number of line buttons supported by a phone type.
type	Assigns the phone type to an SCCP phone.

max-redirect

To change the number of times that a call can be redirected by call forwarding or transfer within a Cisco Unified CME system, use the **max-redirect** command in telephony-service configuration mode. To reset to the default number of redirects, use the **no** form of this command.

max-redirect *number*
no max-redirect

Syntax Description

number	Number of permissible redirects. Range: 5 to 20. Default: 10.

Command Default

Number of redirects is 10.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(24)T1	Cisco Unified CME 7.1	The default value was increased from 5 to 10.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command supports Cisco Unified CME ephone hunt groups by allowing calls to be redirected more than the default number of times.

Examples

The following example sets the maximum number of redirects to 8:

```
Router(config)# telephony-service
Router(config-telephony)# max-redirect 8
```

Related Commands

Command	Description
ephone-hunt	Creates an ephone hunt group in Cisco Unified CME.
hops	Sets the number of hops before a call proceeds to the final number.

max-subscription

To set the maximum number of concurrent watch sessions that are allowed, use the **max-subscription** command in presence configuration mode. To return to the default, use the **no** form of this command.

max-subscription *number*
no max-subscription

Syntax Description

number	Maximum watch sessions. Range: 100 to 500. Default: 100.

Command Default

Maximum subscriptions is 100.

Command Modes

Presence configuration (config-presence)

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command sets the maximum number of concurrent presence subscriptions for both internal and external subscribe requests.

Examples

The following example shows the maximum subscriptions set to 150:

```
Router(config)# presence
Router(config-presence)# max-subscription 150
```

Related Commands

	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
allow subscribe	Allows internal watchers to monitor external presence entities (directory numbers).
presence enable	Allows incoming presence requests from SIP trunks.
server	Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.
watcher all	Allows external watchers to monitor internal presence entities (directory numbers).

max-timeout

To set the maximum combined timeout for the no-answer periods for all ephone-dns in the ephone-hunt list, use the **max-timeout** command in ephone-hunt configuration mode. To return this value to the default, use the **no** form of this command.

max-timeout *seconds*
no max-timeout *seconds*

Syntax Description

seconds	Number of seconds. Range is from 3 to 60000. Default is unlimited.

Command Default

Number of seconds is unlimited.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Examples

The following example shows how to set different no-answer timeouts for each ephone-dn in the hunt-group list and no maximum timeout. The first call to the hunt group rings extension 1001. If that extension does not answer in 7 seconds, the call is forwarded to extension 1002. If that extension does not answer after 10 seconds, the call is forwarded to extension 1003. However, if extension 1003 does not answer after 8 seconds, the call is sent to the final number, extension 4500, because the maximum timeout of 25 seconds has been reached.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 hops 3
 timeout 7, 10, 15
 max-timeout 25
 final 4500
```

Related Commands

	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.

media

To enable media packets to pass directly between the endpoints, without the intervention of the Cisco Unified Border Element (Cisco UBE), and to enable the incoming and outgoing IP-to-IP call gain/loss feature for audio call scoring on either the incoming dial peer or the outgoing dial peer, enter the **media** command in dial peer, voice class, or voice service configuration mode. To return to the default IPIPGW behavior, use the **no** form of this command.

media [{**flow-around** | **flow-through** | **forking** | **monitoring** [*max-calls*] | **statistics** | **transcoder high-density**}]
no media [{**flow-around** | **flow-through** | **forking** | **monitoring** [*max-calls*] | **statistics** | **transcoder high-density**}]

Syntax Description

flow-around	(Optional) Enables media packets to pass directly between the endpoints, without the intervention of the Cisco UBE. The media packet is to flow around the gateway.
flow-through	(Optional) Enables media packets to pass through the endpoints, without the intervention of the Cisco UBE.
forking	(Optional) Enables the media forking feature for all calls.
monitoring	Enables the monitoring feature for all calls or a maximum number of calls.
max-calls	The maximum number of calls that are monitored.
statistics	(Optional) Enables media monitoring.
transcoder high-density	(Optional) Converts media codecs from one voice standard to another to facilitate the interoperability of devices using different media standards.

Command Default

The default behavior of the Cisco UBE is to receive media packets from the inbound call leg, terminate them, and then reoriginate the media stream on an outbound call leg.

Command Modes

Dial peer configuration (config-dial-peer)
Voice class configuration (config-class)
Voice service configuration (config-voi-serv)

Command History

Release	Modification
12.3(1)T	This command was introduced.
12.4(11)XJ2	This command was modified. The **statistics** keyword was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(20)T	This command was modified. The **transcoder** and **high-density** keywords were introduced.
15.0(1)M	This command was modified. The **forking** and **monitoring** keywords and the *max-calls* argument were introduced.

Release	Modification
15.1(3)T	This command was modified. The media flow around is now supported for the SIP to SIP trunk calls in Cisco Unified CME 8.5.

Usage Guidelines

With the default configuration, the Cisco UBE receives media packets from the inbound call leg, terminates them, and then reoriginates the media stream on an outbound call leg. Media flow-around enables media packets to be passed directly between the endpoints, without the intervention of the Cisco UBE. The Cisco UBE continues to handle routing and billing functions. Media flow-around for SIP-to-SIP calls is not supported.

Note The Cisco UBE must be running Cisco IOS Release 12.3(1) or a later release to support media flow-around.

You can specify media flow-around for a voice class, all VoIP calls, or individual dial peers.

The **transcoder high-density** keyword can be enabled in any of the configuration modes with the same command format. If you are configuring the **transcoder high-density** keyword for dial peers, make sure that the **media transcoder high-density** command is configured on both the in and out legs.

The software does not support configuring the **transcoder high-density** keyword on any dial peer that is to handle video calls. The following scenarios are not supported:

- Dial peers used for video at any time. Configuring the **media transcoder high-density** command directly under the dial-peer or a voice-class media configuration is not supported.
- Dial peers configured on a Cisco UBE used for video calls at any time. The global configuration of the **media transcoder high-density** command under voice service voip is not supported.

To enable the **media** command on a Cisco 2900 or Cisco 3900 series Unified Border Element voice gateway, you must first enter the **mode border-element** command. This enables the **media forking** and **media monitoring** commands. Do not configure the **mode border-element** command on the Cisco 2800 or Cisco 3800 series platforms.

Examples

Media Flow-around Examples

The following example shows media flow-around configured on a dial peer:

```
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer) media flow-around
```

The following example shows media flow-around configured for all VoIP calls:

```
Router(config)# voice service voip
Router(config-voi-serv) media flow-around
```

The following example shows media flow-around configured for voice class calls:

```
Router(config)# voice class media 1
Router(config-class) media flow-around
```

Media Flow-though Examples

The following example shows media flow-around configured on a dial peer:

```
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer) media flow-through
```

The following example shows media flow-around configured for all VoIP calls:

```
Router(config)# voice service voip
Router(config-voi-serv) media flow-through
```

The following example shows media flow-around configured for voice class calls:

```
Router(config)# voice class media 2
Router(config-class) media flow-through
```

Media Statistics Examples

The following example shows media monitoring configured for all VoIP calls:

```
Router(config)# voice service voip

Router(config-voi-serv) media statistics
```

The following example shows media monitoring configured for voice class calls:

```
Router(config)# voice class media 1
Router(config-class) media
 statistics
```

Media Transcoder High-density Examples

The following example shows the **media transcoder** keyword configured for all VoIP calls:

```
Router(config)# voice service voip

Router(conf-voi-serv)# media transcoder high-density
```

The following example shows the **media transcoder** keyword configured for voice class calls:

```
Router(config)# voice class media 1
Router(config-voice-class)# media transcoder high-density
```

The following example shows the **media transcoder** keyword configured on a dial peer:

```
Router(config)# dial-peer voice 36 voip
Router(config-dial-peer)# media transcoder high-density
```

Media Monitoring on a Cisco UBE Platform

The following example shows how to configure audio call scoring for a maximum of 100 calls:

```
mode border-element
media monitoring 100
```

Related Commands

Command	Description
dial-peer voice	Enters dial peer configuration mode.
mode border-element	Enables the media monitoring capability of the **media** command.
voice class	Enters voice class configuration mode.
voice service	Enters voice service configuration mode.

members logout

To configure a Cisco Unified CallManager Express system for all non-shared static members or agents in an ephone-hunt with the Hlogout initial state, use the **members logout** command in ephone-hunt configuration mode. To return to the default, use the no form of this command.

This command is not allowed after **list** and **hunt-group logout DND** are configured or if DNs are shared.

members logout
no members logout

Syntax Description

This command has no arguments or keywords.

Command Default

All members are in Hlogin state.

Command Modes

ephone-hunt configuration (config-ephone hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
15.2(4)M	Cisco Unified CME 9.1	This command was introduced.

Usage Guidelines

All members configured under an ephone-hunt are initialized with HLogin. Use this command to initialize all non-shared static members to Hlogout.

Examples

The following example configures HLogout as the default for all non-shared ephone-hunt static members:

```
Router(config-telephony)# ephone-hunt 1
Router(config-ephone-hunt)# members logout
```

Related Commands

Command	Description
ephone-hunt	Enters ephone-hunt configuration mode to define a Cisco CME ephone-hunt group.

members logout (voice hunt-group)

To configure a Cisco Unified CME system for all non-shared static members or agents in a voice hunt group with the Hlogout initial state, use the **members logout** command in voice hunt-group configuration mode. To return to the default state, use the no form of this command.

This command is not allowed if the CLI command **list** is configured .

members logout
no members logout

Syntax Description

This command has no arguments or keywords.

Command Default

All members are in Hlogin state.

Command Modes

voice hunt-group

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1 15.6(3)M1	Cisco Unified CME 11.6	This command was introduced.

Usage Guidelines

All members configured under a voice hunt-group are initialized with HLogin. Use this command to initialize all non-shared static members to Hlogout. If any member of a hunt group in a SIP phone logs out using the CLI command **members logout**, all other DN's of that phone in any hunt group are also logged out. This is because SIP phones only support phone level logout. For SCCP phones, only the DN that is configured with the CLI command **members logout** is logged out from the hunt group. Other member DN's do not logout as SCCP phones support line level logout.

Members Logout is not supported if the CLI command **hunt-group logout DND** is configured. Also, you cannot configure the CLI command **members logout** if the command **list** is configured.

Examples

The following example configures HLogout as the default for all non-shared voice hunt-group static members:

```
Router(config-register-global)# voice hunt-group 1
Router(config-voice-hunt-group)# members logout
```

Related Commands

Command	Description
members logout	Enables members logout for ephone-hunt groups configured on a Cisco Unified CME.

missed-calls

To report missed calls to directory numbers on an IP phone, use the **missed-calls** command in ephone configuration mode. To suppress missed-calls reporting, use the **no** form of this command.

missed-calls [**all**]
no missed-calls

Syntax Description

all	(Optional) Displays all missed calls including those on overlay buttons.

Command Default

Missed calls are presented on the IP phone and listed in the missed-calls directory. Missed calls to overlay buttons are not reported.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command enables Cisco Unified CME to report missed calls on the specified phone. Use the **all** keyword to report missed calls to overlaid directory numbers. Only calls to an overlay set that are visibly presented on the phone are reported as missed calls. Calls to an overlay that are terminated by the caller before they are displayed on the phone are not reported as missed calls.

If the unique extension number for a phone is assigned to an overlay set on the phone, missed calls to that extension number are not reported unless you enable this command using the **all** keyword.

Examples

The following example shows that all unanswered calls to 4001 are reported on phone 1.

```
ephone-dn  1  dual-line
 number 4001

ephone  1
 mac-address 0014.6AAC.24E3
 type 7960
 button  1o1,30,31 2:2 3:3
 missed-calls all
```

Related Commands

Command	Description
button	Associates directory numbers with individual buttons on a Cisco Unified IP Phone and specifies ring behavior.

mlpp indication

To enable MLPP indication on an SCCP phone or analog FXS port, use the **mlpp indication** command in ephone-template or voice-port configuration mode. To disable MLPP indication, use the **no** form of this command.

mlpp indication
no mlpp indication

Syntax Description

This command has no arguments or keywords.

Command Default

MLPP indication is enabled on the phone.

Command Modes

Ephone-template configuration (config-ephone-template)
Voice-port configuration (config-voiceport)

Command History

Cisco IOS Release	Cisco Products	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

This command enables a phone to play precedence and preemption tones, and display precedence information for calls. If MLPP indication is disabled, calls on the phone can be preempted but there is no visual or audible indication.

To apply a template to an SCCP phone, use the **ephone-template** command in ephone configuration mode.

Examples

The following example shows MLPP indication is disabled in template 5 and applied to phone 12:

```
ephone-template  5
 mlpp max-precedence 0
 no mlpp indication
!
!
ephone  12
 mac-address 000F.9054.31BD
 ephone-template 5
 type 7960
 button  1:12
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an SCCP phone.
mlpp max-precedence	Sets the maximum precedence (priority) level that a phone user can specify when making an MLPP call.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.

Command	Description
preemption tone timer	Sets the amount of time the preemption tone plays on the called phone when a lower precedence call is being preempted.

mlpp max-precedence

To set the maximum precedence (priority) level that a phone user can specify when making an MLPP call, use the **mlpp max-precedence** command in ephone-template or voice-port configuration mode. To reset to the default, use the **no** form of this command.

mlpp max-precedence *number*
no mlpp max-precedence

Syntax Description

number	Number representing the maximum precedence level. Range: 0 to 4, where 0 is the highest priority. Default: 4.

Command Default

The MLPP precedence is 4 (routine).

Command Modes

Ephone-template configuration (config-ephone-template)
Voice-port configuration (config-voiceport)

Command History

Cisco IOS Release	Cisco Products	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

This command sets the maximum precedence level that a user can select when making MLPP calls from a phone. The phone user can specify a precedence level that is less than or equal to this value. Cisco Unified CME rejects the call if a user selects a precedence level that is higher than the level set with this command and the user receives an error tone.

Emergency 911 calls are automatically assigned precedence level 0.

To apply a template to an SCCP phone, use the **ephone-template** command.

Examples

The following example shows the precedence level set to 0 in template 5 and applied to phone 12:

```
ephone-template  5
 mlpp max-precedence 0
!
!
ephone  12
 mac-address 000F.9054.31BD
 ephone-template 5
 type 7960
 button  1:12
```

Related Commands

Command	Description
access-digits	Defines the access digit that phone users dial to request a precedence call.
ephone-template (ephone)	Applies an ephone template to an SCCP phone.

Command	Description
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
mlpp preemption	Enables the preemption capability on an SCCP phone or analog FXS port.

mlpp preemption

To enable calls on an SCCP phone or analog FXS port to be preempted, use the **mlpp preemption** command in ephone-template or voice-port configuration mode. To disable preemption, use the **no** form of this command.

mlpp preemption
no mlpp preemption

Syntax Description

This command has no arguments or keywords.

Command Default

Preemption is enabled on the phone.

Command Modes

Ephone-template configuration (config-ephone-template)
Voice-port configuration (config-voiceport)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

The command allows an SCCP IP phone or an FXS analog phone to have its calls preempted if it is busy with lower precedence calls.

A phone with preemption disabled can still receive precedence calls in an MLPP network, but the phone itself does not get preempted. The preemption-disabled phone can be connected to a call that is preempted (at another device), in which case, that device receives preemption.

To apply a template to an SCCP phone, use the **ephone-template** command in ephone configuration mode.

Examples

The following example shows preemption disabled in template 5 and applied to phone 12:

```
ephone-template  5
 mlpp max-precedence 0
 no mlpp preemption
!
!
ephone  12
 mac-address 000F.9054.31BD
 ephone-template 5
 type 7960
 button  1:12
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an SCCP phone.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.

Command	Description
mlpp max-precedence	Sets the maximum precedence (priority) level that a phone user can specify when making an MLPP call.
preemption tone timer	Defines the expiry time for the preemption tone for the call being preempted.

mlpp service-domain

To set the service domain and maximum precedence (priority) level for Multilevel Precedence and Preemption (MLPP) calls, use the **mlpp service-domain** command in ephone-template or voice-port configuration mode. To reset to the default, use the **no** form of this command.

mlpp service-domain {**drsn** | **dsn**} **identifier** *domain-number* **max-precedence** *level*
no mlpp service-domain

Syntax Description

drsn	Phone belongs to Defense Red Switched Network (DRSN).
dsn	Phone belongs to Defense Switched Network (DSN).
domain-number	Number to identify the domain, in three-octet format. Range: 0x000000 to 0xFFFFFF.
level	Number representing the maximum precedence level, where 0 is the highest priority. Range is 0 to 4 (DSN) or 0 to 5 (DRSN).

Command Default

Phone uses global default configured with the **service-domain** command.

Command Modes

Ephone-template configuration (config-ephone-template)
Voice-port configuration (config-voiceport)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command sets the MLPP domain type and number for the phone, and the maximum precedence level that a user can select when making MLPP calls from the phone.

The phone user can select a precedence level that is less than or equal to the value set with this command. Cisco Unified CME rejects the call if a user selects a precedence level that is higher than the level set with this command and the user receives an error tone.

If this command and the **service-domain** command are not enabled, the phone cannot make MLPP calls.

Emergency 911 calls are automatically assigned precedence level 0.

To apply a template to an SCCP phone, use the **ephone-template** command.

Examples

The following example shows the precedence level set to 1 in template 5 and applied to phone 15:

```
ephone-template  5
 mlpp service-domain dsn identifier 000010 max-precedence 1
!
!
ephone  15
 mac-address 000F.9054.31BD
 ephone-template 5
```

```
type 7960
button  1:15
```

Related Commands

Command	Description
access-digit	Defines the access digit that phone users dial to request a precedence call.
ephone-template (ephone)	Applies an ephone template to an SCCP phone.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
mlpp preemption	Enables the preemption capability on an SCCP phone or analog FXS port.
service-domain	Sets the global MLPP domain name and number.

mobility (ephone-dn)

To enable the Mobility feature on an extension of an SCCP IP phone, use the **mobility** command in ephone-dn configuration mode. To disable mobility on the extension, use the **no** form of this command.

mobility
no mobility

Syntax Description

This command has no arguments or keywords.

Command Default

Mobility is not enabled for the extension.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command enables the Mobility feature on the extension, which is required to enable the Single Number Reach (SNR) feature.

Examples

The following example shows extension 1001 is enabled for SNR. After a call rings at this number for 5 seconds, the call also rings at the remote number 4085550133. If the call is not answered after 20 seconds, the call no longer rings the phone and is forwarded to the voice-mail number 2001.

```
ephone-dn 10
 number 1001
 mobility
 snr 4085550133 delay 5 timeout 15 cfwd-noan 2001
```

Related Commands

Command	Description
number	Associates a telephone or extension number with an ephone-dn.
snr	Enables Single Number Reach on an extension of an SCCP IP phone.
softkeys connected	Modifies the order and type of soft keys that display on an IP phone during the connected call state.
softkeys idle	Modifies the order and type of soft keys that display on an IP phone during the idle call state.

mobility (voice register dn)

To enable the Mobility feature on an extension of a Cisco Unified SIP IP phone, use the **mobility** command in voice register dn configuration mode. To disable the Mobility feature on the extension, use the **no** form of this command.

mobility
no mobility

Syntax Description

This command has no arguments or keywords.

Command Default

The Mobility feature is not enabled on the extension of a Cisco Unified SIP IP phone.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **mobility** command to enable a Cisco Unified SIP IP phone to receive calls on an extension, which is required to enable the Single Number Reach (SNR) feature.

Examples

The following example shows how to enable the Mobility feature on directory number 25:

```
Router(config)# voice register dn 25
Router(config-register-dn)# mobility
```

Related Commands

Command	Description
snr (voice register dn)	Enables the SNR feature on an extension of a Cisco Unified SIP IP phone.

mode

The **mode** command under voice register global allows you to select the operating mode for Cisco Unified Call Manager Express and Cisco Unified Survivable Remote Site Telephony. The default operation for voice register global is SRST mode, which may be selected using the **no** form of this command.

mode [cme | cme-app | esrst]
no mode

Syntax Description

cme	CME mode provides the options to configure SIP phones and features.
cme-app	CME appliance mode provides the options to configure up to 200 SIP phones for routers that are only being used to provide call control. CME appliance mode is available only for ISR4321 routers.
esrst	E-SRST mode extends SRST features to include the static configuration of SIP phones and features.

Command Default

Default is SIP SRST mode.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST	This command was introduced.
15.3(3)M	Cisco CME 10.0	This command was modified to add the **esrst** mode.
Cisco IOS XE Everest 16.5.1b	Unified CME 11.7 Unified SRST 11.7	The behavior of **no** form of this command was modified, to clear all voice register pools and voice register dns, along with mode-specific configurations.
Cisco IOS XE Amsterdam 17.2.1r	Unified CME 12.8	The **cme-app** mode was added for ISR4321 routers, allowing the configuration of up to 200 phones (For routers dedicated to CME use only).

Usage Guidelines

The mode command selects the SIP call control application to be used and enables associated configuration commands. By default, the router operates in SRST mode, allowing only phones failing over from Unified Communications Manager to register. Enter **cme** or **cme-app** modes to configure SIP phones and features for standalone call control use.

E-SRST mode provides a hybrid of these applications, where CME features may be configured to complement the SRST application.

For releases prior to Unified CME/SRST 11.7, the **no** form of this command clears only the mode-specific configurations (For example, **source-address** under voice register global configuration, and user credentials configured under voice register pool configuration). From Cisco IOS XE Everest 16.5.1 (Unified CME/SRST Release 11.7) onwards, the **no** form of this command clears all the voice register pools and voice register dns, along with mode-specific configurations.

Examples

The following example shows how to set the mode to Cisco CME:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
```

Related Commands

	Description
show voice register global	Displays all global configuration information that is associated with SIP phones.

moh (ephone-dn)

To enable music on hold (MOH) from an external live audio feed (standard line-level audio connection) connected directly to the router by an foreign office exchange (FXO) or an E&M analog voice port, use the **moh** command in ephone-dn configuration mode. To disable MOH from a live feed or to disable the outcall number or multicast capability, use the **no** form of this command.

moh [**out-call** *outcall-number*] [**ip** *ip-address* **port** *port-number* [**route** *ip-address*]]
no **moh** [{**out-call** *outcall-number* | **ip**}]

Syntax Description

out-call *outcall-number*	(Optional) Sets up a call to the outcall number in order to connect to the MOH feed. If this keyword is not used, the live feed is assumed to derive from an incoming call to the ephone-dn under which this command is used.
ip *ip-address*	(Optional) Indicates that this audio stream is to be used as a multicast source as well as the MOH source and specifies the destination IP address for multicast.
port *port-number*	(Optional) Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CallManager Express router.
route *ip-address*	(Optional) Indicates the specific router interface on which to transmit the IP multicast packets. The default is that the MOH multicast stream is automatically output on the interface that corresponds to the address that was configured with the **ip source-address** command.

Command Default

MOH is disabled on an extension.

Command Modes

Ephone-dn configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The **ip**, **port**, and **route** keywords were added.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command takes the specified live-feed audio stream and uses it as MOH for a Cisco Unified CME system. The connection for the live-feed audio stream is established as an automatically connected voice call. If the **out-call** keyword is used, the type of connection can include VoIP calls if voice activity detection (VAD) is disabled. The typical operation is for the MOH ephone-dn to establish a call to a local router E&M voice port.

Connection via E&M is the recommended mechanism because it requires minimal external components. The E&M port must be placed in 4-wire operation, using E&M immediate signaling and with the **auto-cut-through** option enabled. You directly connect a line-level audio feed (standard audio jack) to pins 3 and 6 of an E&M RJ-45 connector. The E&M WAN interface card (WIC) has a built-in audio transformer that provides appropriate electrical isolation for the external audio source. (The audio connection on the E&M port does not require loop current.) The **signal immediate** and **auto-cut-through** commands disable E&M signaling on this voice port. A G.711 audio packet stream is generated by the digital signal processor (DSP) on the E&M port.

If you are using an FXO voice port for live-feed MOH instead of an E&M port, connect the MOH source to the FXO voice port. This connection requires an external adapter to supply normal telephone company (telco) battery voltage with the correct polarity to the tip-and-ring leads of the FXO port. The adapter must also provide transformer-based isolation between the external audio source and the tip-and-ring leads of the FXO port.

Music from a live feed is continuously fed into the MOH playout buffer instead of being read from an audio file in flash memory. There is typically a two-second delay with live-feed MOH.

If the **out-call** keyword is used, an outbound call to the MOH live-feed source is attempted (or reattempted) every 30 seconds until the call is connected to the ephone-dn (extension) that has been configured for MOH. Note that this ephone-dn is not associated with any physical phone.

If the **moh** (ephone-dn) command is used without any keywords or arguments, the ephone-dn will accept an incoming call and use the audio stream from the call as the source for the MOH stream, displacing any audio stream that is available from a flash file. To accept an incoming call, the ephone-dn must have an extension or phone number configured for it. A typical usage would be for an external H.323-based server device to call the ephone-dn to deliver an audio stream to the Cisco CME system. Normally, only a single ephone-dn would be configured like this. If there is more than one ephone configured to accept incoming calls for MOH, the first ephone-dn that is successfully connected to a call (incoming or outgoing) is the MOH source for the system.

MOH can also be derived from an audio file when you use the **moh** command in telephony-service configuration mode with the *filename* argument. There can be only one MOH stream at a time in a Cisco CME system, and if both an audio file and a live feed have been specified for the MOH stream, the router seeks the live feed from the **moh (ephone-dn)** command first. If the live feed is found, the router displaces the audio file source. If the live feed is not found or fails at any time, the router falls back to the audio file source that was specified in the **moh (telephony-service)** command.

If you use the **ip** keyword to specify a multicast address in this command, the audio stream is sent to the multicast address in addition to serving as the MOH source. Additionally, if you specify a different multicast address using the **multicast moh** command under telephony-service configuration mode, the audio stream is also sent to the multicast address that you named in that command. It is therefore possible to send the live-feed audio stream to MOH and to two different multicast addresses: the one that is directly configured under the **moh (ephone-dn)** command and the one that is indirectly configured under the **multicast moh** (telephony-service) command.

A related command, the **feed** command, provides the ability to multicast an audio stream that is not the MOH audio stream.

Note IP phones do not support multicast at 224.x.x.x addresses.

Examples

The following example establishes a live music-on-hold source by setting up a call to extension 7777:

```
Router(config)# ephone-dn 55
Router(config-ephone-dn)# moh out-call 7777
```

Related Commands

	Description
auto-cut-through	Enables call completion when an M-lead response is not provided.
ephone-dn	Enters ephone-dn configuration mode to set directory numbers and parameters for individual Cisco IP phone extensions.
feed	Enables multicast of an audio stream that is different from the music-on-hold audio stream.
ip source-address	Identifies the IP address and port through which IP phones communicate with a Cisco CME router.
moh (telephony-service)	Enables music on hold from an audio file.
multicast moh	Enables multicast of the music-on-hold audio stream.
signal	Specifies the type of signaling for a voice port.

moh (telephony-service)

To generate an audio stream from a file for music on hold (MOH) in a Cisco CallManager Express (Cisco CME) system, use the **moh** command in telephony-service configuration mode. To disable the MOH audio stream from this file, use the **no** form of this command.

moh *filename*
no moh

Syntax Description

filename	Name of the audio file to use for the MOH audio stream. The file must be copied to flash memory on the Cisco CME router.

Command Default

Tone on hold (a periodic beep is played to the caller)

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

This command enables MOH from .au and .wav format music files. MOH is played for G.711 callers and on-net VoIP and PSTN callers who are on hold in a Cisco CME system. Local callers within a Cisco CME system hear a repeating tone while they are on hold.

Audio files that are used for MOH must be copied to the Cisco CME router flash memory. A MOH file can be in .au or .wav file format; however, the file format must contain 8-bit 8-kHz data in A-law or mu-law data format. We recommend using a moh-file size greater than 100 KB.

If you want to replace or modify the audio file that is currently specified, you must first disable the MOH capability using the **no moh** command. The following example replaces file1 with file2:

```
Router(config-telephony)# moh file1
Router(config-telephony)# no moh
Router(config-telephony)# moh file2
```

If you specify a second file without first removing the original file, the MOH mechanism stops working and may require a router reboot to clear the problem.

A related command, the **moh** command in ephone-dn configuration mode, can be used to establish a MOH audio stream from a live feed. If you configure both commands, MOH falls back to playing music from the audio file if the live music feed is interrupted.

The **multicast moh** command allows you to use the MOH stream for a multicast broadcast.

When the **multicast moh** and **debug ephone moh** commands are both enabled, if you also use the **no moh** command, the debug output can be excessive and flood the console. Multicast MOH should be disabled before using the **no moh** command when the **debug ephone moh** command is enabled.

Examples

The following example enables music on hold and specifies a music file:

```
Router(config)# telephony-service
Router(config-telephony)# moh minuet.wav
```

Related Commands

	Description
debug ephone moh	Displays diagnostic information for music on hold.
moh (ephone-dn)	Enables music on hold from a live audio feed.
multicast moh	Enables multicast of the music-on-hold audio stream.

moh (voice moh-group)

To enable music on hold (MOH) for a MOH group, use the **moh** command in voice moh-group configuration mode. To disable music on hold, use the no form of this command.

moh *filename*
no moh *filename*

Syntax Description

filename	Name of the music file. The music file must be in the system flash.

Command Default

No MOH is enabled

Command Modes

Voice moh-group configuration (config-voice-moh-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME/SRST/SIP SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME/SRST/SIP SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

The **moh** command allows you to specify the .au and .wav format music files that are played to callers who have been put on hold. MOH works only for G.711 calls and on-net VoIP and PSTN calls. For all other calls, callers hear a periodic tone. You must provide the directory and filename of the MOH file in URL format. For example: moh flash:/minuet.au

Note Music-on-hold files can be in .wav or .au file format; however, the file format must contain 8-bit 8-kHz data; for example, CCITT a-law or u-law data format.

Examples

The following example enables MOH for voice moh group 1 and specifies the music files:

```
Router(config)#
Router(config)#voice moh-group 1
Router(config-voice-moh-group)# moh flash:/minuet.wav
```

Related Commands

voice moh-group	Enters voice moh-group configuration mode.
extension-range	Defines extension range for a clients calling a voice-moh-group.
moh	Enables music on hold from a flash audio file.
multicast moh	Enables multicast of the music-on-hold audio stream.

moh-file-buffer

To specify a MOH file buffer size, use the **moh-file-buffer** command in telephony-service configuration mode. To delete the moh-file-buffer size, use the **no** form of this command.

moh-file-buffer *file-size*
no moh-file-buffer

Syntax Description

file-size	Specifies a numeric value for the buffer MOH file size between 64 KB and 10000 KB.

Command Default

No moh-file-buffer is configured.

Command Modes

Telephony-service configuration (config-telephony-service)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command allows to set a buffer MOH file size limit for new MOH files. You can allocate a MOH file buffer size between 64 KB (8 seconds) and 10000 KB (20 minutes, approximately). A large buffer size is desirable to cache the largest MOH file and better MOH performance. During memory allocation the buffer size is aligned to 16KB.

The default maximum file buffer size is 64 KB. If the MOH file size is too large, it cannot be cached and the buffer size falls back to 64 KB.

Note When live-feed is enabled there is no file caching for MOH-group 0.

Examples

The following example shows a moh-file-buffer size of 180 KB assigned for future moh files under the telephony-service configuration mode.

```
!
!
!
telephony service
 max-conferences 8 gain -6
 transfer-system full-consult
 moh-file-buffer 180
!
!
line con 0
 exec-timeout 0 0
line aux 0
```

Related Commands

Command	Description
voice-moh-group	Enter voice-moh-group configuration mode.
moh filename	Enables music on hold from a flash audio feed
multicast moh	Enables multicast of the music-on-hold audio stream.
extension-range	Specifies the extension range for a clients calling a voice-moh-group.

moh-group (ephone-dn)

To assign a MOH group to a directory number, use the **moh-group** command in ephone-dn configuration mode. To remove the MOH group, use the **no** form of this command.

moh-group *tag*
no moh-group *tag*

Syntax Description

tag	A unique number that identifies a MOH group. Range is from 1 to 5.

Command Default

No MOH group is configured.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use this command to assign a MOH group to a directory number in ephone-dn configuration mode. Use the number tag from 1 to 5 to specify the MOH group that you want to assign to this directory number.

Examples

The following example shows how to assign a MOH group to a directory number under ephone-dn mode.

```
Router(config)# ephone-dn 98
Router(config-ephone-dn)#moh-group 1
Router(config-ephone-dn)#
```

Related Commands

description	Displays a brief description about a voice moh-group in use.
extension-range	Defines extension range for a clients calling a voice-moh-group.
moh	Enables music on hold from a flash audio feed.
multicast moh	Enables multicast of the music-on-hold audio stream.

mtp

To send voice packets from an IP phone to the Cisco Unified CME router, use the **mtp** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

mtp { **video-only** | **both** }
no **mtp** { **video-only** | **both** }

Syntax Description

video-only	Specifies that the video streams must be sent through the Cisco Unified CME route.
both	Specifies that both voice and video streams must be sent through the Cisco Unified CME route.

Command Default

If no arguments are given, only voice packets are sent to the router.

An IP phone in a call with another IP phone in the same Cisco Unified CME system sends voice and video packets directly to the other phone.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
15.1(4)M	Cisco Unified CME 8.6	Support for choosing video streams was added.

Usage Guidelines

Normally, media packets (RTP packets) that are sent between IP phones in the same Cisco Unified CME system go directly to the other phone and do not travel through the Cisco Unified CME router. When these packets are sent from a remote IP phone to another IP phone in the same Cisco Unified CME system, they may be obstructed by a firewall. The **mtp** command instructs a phone to always send its media packets to the Cisco Unified CME router, which acts as a proxy and forwards the packets to the destination. Firewalls can then be easily configured to pass the RTP packets because the router uses a specified UDP port for media packets. In this way, RTP packets from remote IP phones can be delivered to IP phones on the same system. The default is that this function is off and that RTP packets that are sent from one IP phone to another IP phone in the same Cisco Unified CME system go directly to the other phone.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example sends video and audio packets from ephone 437 to the Cisco Unified CME router for all calls:

```
ephone 437
 button 1:29
 mtp both
```

Related Commands

Command	Description
ephone-template (ephone)	Applies template to ephone being configured.

mtu (vpn-profile)

To enter the mtu value in bytes, use the **mtu** command in vpn-profile configuration mode.

mtu bytes

Syntax Description

bytes	Mtu value, in bytes. Range: 256 to 1406. Default: 1290.

Command Default

Default is 1290 bytes.

Command Modes

Vpn-profile configuration (conf-vpn-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use the mtu command to define a value in bytes.. The mtu value ranges from 256 to 1406 bytes. The defaul value is 1290 bytes.

Examples

The following example shows the mtu value set to 1300 bytes in vpn-profile 2:

```
Router# show run
!
!
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  keepalive 50
  host-id-check disable
 vpn-profile 2
  mtu 1300
  password-persistent enable
  host-id-check enable
 sip
!
voice class media 10
 media flow-around
!
!
voice register global
 max-pool 10
```

Related Commands

Command	Description
vpn-profile	Defines a VPN-profile.

multicast moh

To use the music-on-hold (MOH) audio stream as a multicast source for Cisco Unified CME or for a MOH group, use the **multicast moh** command in telephony-service configuration mode or in voice-moh-group configuration modep. To disable multicast use of the MOH stream, use the **no** form of this command.

multicast moh *ip-address* **port** *port-number* [**route** *ip-address-list*]
no multicast moh

Syntax Description

ip-address	Specifies the destination IP address for multicast.
port *port-number*	Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CME router.
route *ip-address-list*	(Optional) Indicates specific router interfaces over which to transmit the IP multicast packets. Up to four IP addresses can be listed, each separated from the other by a space. The default is that the MOH multicast stream is automatically output on the interfaces that correspond to the address that was configured with the **ip source-address** command.

Command Default

No multicast is enabled.

Command Modes

Telephony-service configuration (config-telephony)
Voice-moh-group configuration (config-voice-moh-group)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. Multicast MOH was enabled under voice moh-group configuration mode.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command enables multicast of the audio stream that is designated for MOH in a Cisco CME system. Use this command in voice moh-group configuration mode to enable multicast of audio stream for a specific MOH group.

A related command, the **moh (ephone-dn)** command, creates a MOH audio stream from an external live feed and optionally enables multicast on that stream. These two commands can be used concurrently to provide multicast of a live-feed MOH audio stream to two different multicast addresses.

Another related command, the **feed** command, enables multicast of an audio stream that is not the MOH audio stream.

When the **multicast moh** and **debug ephone moh** commands are both enabled, if you also use the **no moh** command, the debug output can be excessive and flood the console. Multicast MOH should be disabled before using the **no moh** command when the **debug ephone moh** command is enabled.

Note IP phones do not support multicast at 224.x.x.x addresses.

Note Multicast for live feed is not support in MOH groups.

Examples

The following example enables multicast of the MOH audio stream at multicast address 239.10.16.4 and names two router interfaces over which to send the multicast packets.

Example 1: Multicast enabled for MOH audio stream under telephony service.

```
Router(config)# telephony-service
Router(config-telephony)# moh minuet.au
Router(config-telephony)# multicast moh 239.10.16.4 port 2000 route 10.10.29.17 10.10.29.33
```

Example 2: Multicast enabled for MOH audio stream under voice moh-group configuration mode.

```
Router(config)# voice-moh-group 1
Router(config-voice-moh-group)# moh minuet.au
Router(config-voice-moh-group)# multicast moh 239.10.16.4 port 2000 route 10.10.29.17
10.10.29.33
```

Related Commands

Command	Description
feed	Enables multicast of an audio stream that is not the music-on-hold audio stream.
ip source-address	Identifies the IP address and port through which IP phones communicate with a Cisco CME router.
moh (ephone-dn)	Enables music on hold from a live audio feed.
moh (telephony-service)	Enables music on hold from an audio file.
voice moh-group	Enters voice moh-group configuration mode

mwi (ephone-dn and ephone-dn-template)

To enable a specific Cisco Unified IP phone extension to receive message-waiting indication (MWI) notification from an external voice-messaging system, use the **mwi** command in ephone-dn or ephone-dn-template configuration mode. To disable this feature, use the **no** form of this command.

mwi {**off** | **on** | **on-off**}
no mwi {**off** | **on** | **on-off**}

Syntax Description

off	Sets a Cisco Unified IP phone extension to process MWI to OFF, using either the main or secondary phone number.
on	Sets a Cisco Unified IP phone extension to process MWI to ON, using either the main or secondary phone number.
on-off	Sets a Cisco Unified IP phone extension to process MWI to both ON and OFF, using either the main or secondary phone number.

Command Default

MWI notification is disabled on an extension.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command enables a Cisco Unified IP phone extension to receive MWI notification from an external voice-messaging system for all the Cisco Unified IP phones connected to the Cisco Unified CME router. This extension is a “dummy” extension and is not associated with any physical phone. The external voice-messaging system is able to communicate MWI status by making telephone calls to the dummy extension number, with the MWI information embedded in either the called or calling-party IP phone number.

This command cannot be used unless the **number** command is already configured for this extension (ephone-dn).

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example sets MWI to on:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number 8000
Router(config-ephone-dn) mwi on
```

The following example sets MWI to off:

```
Router(config)# ephone-dn 2
Router(config-ephone-dn) number 8001
Router(config-ephone-dn) mwi off
```

The following example sets MWI to both on and off for the primary and secondary number, where the MWI information is embedded in the calling-party number. A call placed by the voice-mail system to 8002 turns on the MWI light for the extension number indicated by the calling-party number for the MWI call. A call placed to 8003 turns off the MWI light.

```
Router(config)# ephone-dn 3
Router(config-ephone-dn) number 8002 secondary 8003
Router(config-ephone-dn) mwi on-off
```

The following example sets MWI to both on and off for the primary and secondary number, where the MWI information is embedded in the called-party number. A call placed by the voice-mail system to 8000*5001*1 turns on the MWI light for extension 5001. A call placed to 8000*5001*2 turns off the MWI light.

```
Router(config)# ephone-dn 20
Router(config-ephone-dn) number 8000*....*1 secondary 8000*....*2
Router(config-ephone-dn) mwi on-off
```

The following example uses an ephone-dn-template to set MWI to on:

```
Router(config)# ephone-dn-template 4
Router(config-ephone-dn-template) mwi on
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 8000
Router(config-ephone-dn)# ephone-dn-template 4
```

Related Commands

	Description
ephone-dn-template (ephone-dn)	Applies template to ephone-dn being configured.
mwi expires	Sets the expiration timer for registration for either the client or the server.
mwi sip (ephone-dn)	Subscribes an extension in a Cisco Unified CME router to receive MWI notification from a SIP MWI server.
mwi sip-server (telephony-service)	Configures the IP address and port number for an external SIP-based MWI server.
number	Associates a telephone or extension number with an extension (ephone-dn) in a Cisco Unified CME system.

mwi (voice register dn)

To enable a specific Cisco IP phone extension (directory number) associated with a SIP phone to receive message-waiting indication (MWI) notification, use the **mwi** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

mwi
no mwi

Syntax Description

This command has no arguments or keywords.

Command Default

This command is disabled

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command enables a particular extension on a SIP IP phone to receive MWI notification.

For Cisco Unified CME 4.1 and later versions, MWI requires that SIP phones must be configured with a directory number by using the **number** (voice register pool) command with the dn keyword; direct line numbers are not supported.

Examples

The following example shows how to enable MWI for a particular extension number associated with a SIP IP phone:

```
Router(config)# voice register dn 4
Router(config-register-dn)# mwi
```

Related Commands

	Description
number (voice register pool)	Configures number patterns for a voice register pool.

mwi expires

To set the expiration timer for registration for the message-waiting indication (MWI) client or server, use the **mwi expires** command in telephony-service configuration mode. To disable the timer, use the **no** form of this command.

mwi expires *seconds*
no mwi expires *seconds*

Syntax Description

seconds	Expiration time, in seconds. Range is from 600 to 99999. Default is 86400 (24 hours).

Command Default

Default is 86400 seconds (24 hours).

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Examples

The following example sets the expiration timer to 1000 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# mwi expires 1000
```

Related Commands

	Description
mwi relay (telephony-service)	Enables the Cisco CME router to relay MWI information to remote Cisco IP phones.
mwi sip-server (telephony-service)	Configures the IP address and port number for the external SIP-based MWI server.

mwi prefix

To specify a prefix for an extension that will receive unsolicited message-waiting indication (MWI) from an external SIP-based MWI server, use the **mwi prefix** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

mwi prefix *prefix-string*
no mwi prefix

Syntax Description

prefix-string	Digits at the beginning of a number that will be recognized as a prefix before a Cisco Unified CME extension number. The maximum prefix length is 32 digits.

Command Default

A prefix is not defined.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Central voice-messaging servers that provide mailboxes for several Cisco Unified CME sites may use site codes or prefixes to distinguish among similarly numbered ranges of extensions at different sites. In Cisco Unified CME 4.0 and later versions, you can specify that your Cisco Unified CME system should accept unsolicited SIP Notify messages for message-waiting indication (MWI) that include a prefix string as a site identifier.

For example, an MWI message might indicate that the central mailbox number 5551234 has a voice message. In this example, the digits 555 are set as the prefix string or site identifier using the **mwi prefix** command. The local Cisco Unified CME system is able to convert 5551234 to 1234 and deliver the MWI to the correct phone. Without this prefix string manipulation, the system would reject an MWI indication for 5551234 as not matching the local Cisco Unified CME extension 1234.

Examples

The following example identifies the SIP server for MWI notification at the IP address 172.16.14.22. It states that the Cisco Unified CME system will accept unsolicited SIP Notify messages for known mailbox numbers (extension numbers) that are prefixed with the digits 555.

```
sip-ua
 mwi-server 172.16.14.22 unsolicited
telephony-service
 mwi prefix 555
```

Related Commands

	Description
mwi (ephone-dn and ephone-dn-template)	Configures specific Cisco Unified IP phone directory numbers to receive MWI notification from an external voice-mail system.

	Description
mwi-server	Configures MWI server parameters.
mwi sip (ephone-dn)	Subscribes an extension in a Cisco Unified CME router to receive MWI notification from a SIP MWI server.

mwi qsig

To enable Cisco Unified CME to interrogate a QSIG message center for the message-waiting indication (MWI) status of an IP phone extension, use the **mwi qsig** command in ephone-dn or ephone-dn-template configuration mode. To return to the default, use the **no** form of this command.

mwi qsig
no mwi qsig

Syntax Description

This command has no arguments or keywords.

Command Default

An extension is not subscribed to receive MWI using QSIG.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **transfer-system** command must be used with the **full-consult** or **full-blind** keyword to enable H.450 call forwarding.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

In the following example, a voice mail extension (7000) and a normal extension (7582) are defined. Calls are forwarded to voice mail when extension 7582 is busy or does not answer. The message-waiting indicator (MWI) on extension 7582's phone is subscribed to receive notifications from the QSIG message center.

```
ephone-dn 25
 number 7582
 mwi qsig
 call-forward busy 7000
 call-forward noan 7000 timeout 20
telephony-service
 voicemail 7000
 transfer-system full-consult
```

Related Commands

	Description
ephone-dn-template (ephone-dn)	Applies a template to ephone-dn being configured.
transfer-system	Specifies the call transfer method for Cisco Unified CME extensions.

	Description
voicemail	Defines the telephone number that is speed-dialed when the Messages button on a Cisco Unified IP phone is pressed.

mwi reg-e164

To register E.164 numbers rather than extension numbers with a Session Interface Protocol (SIP) proxy or registrar, use the **mwi reg-e164** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

mwi reg-e164
no mwi reg-e164

Syntax Description

This command has no keywords or arguments.

Command Default

Registering extension numbers with the SIP proxy or registrar.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T7 12.4	Cisco CME 3.3	This command was introduced.

Usage Guidelines

This command is used when setting up extensions to use an external SIP-based message-waiting indication (MWI) server. The **mwi-server** command in SIP user-agent configuration mode specifies other settings for MWI service.

Examples

The following example specifies that E.164 numbers should be used for registration with the SIP proxy or registrar:

```
telephony-service
 mwi reg-e164
```

Related Commands

	Description
mwi-server (SIP user-agent)	Specifies voice-mail server settings on a voice gateway or user agent (UA).

mwi relay

To enable a Cisco CallManager Express (Cisco CME) router to relay message-waiting indication (MWI) notification to remote Cisco IP phones, use the **mwi relay** command in telephony-service configuration mode. To disable MWI relay, use the **no** form of this command.

mwi relay
no mwi relay

Syntax Description

This command has no arguments or keywords.

Command Default

MWI relay is disabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced command.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T command.

Usage Guidelines

Use this command to enable the Cisco CME router to relay MWI notification to remote Cisco IP phones. The router at the central site acts as a notifier after this command is used.

Examples

The following example enables MWI relay:

```
Router(config)# telephony-service
Router(config-telephony)# mwi relay
```

Related Commands

	Description
mwi expires	Sets the expiration timer for registration for the client or the server.
show mwi relay clients	Displays registration information for MWI relay clients.

mwi sip

To subscribe an extension in a Cisco Unified CME system to receive message-waiting indication (MWI) from a SIP-based MWI server, use the **mwi sip** command in ephone-dn or ephone-dn-template configuration mode. To remove the configuration, use the **no** form of this command.

mwi sip
no mwi sip

Syntax Description

This command has no arguments or keywords.

Command Default

An extension is not subscribed to receive MWI.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to subscribe an extension in a Cisco Unified CME router to receive MWI notification from a SIP-based MWI server, and use the **mwi sip-server** command to specify the IP address and port number for the external SIP-based MWI server. This function integrates a Cisco Unified CME router with a SIP-protocol-based MWI service.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example subscribes extension 5001 to receive MWI notification from an external Session Initiation Protocol (SIP) MWI server and requests the SIP MWI server to send MWI notification messages through SIP to the Cisco Unified CME router for extension 5001:

```
Router(config) ephone-dn 1
Router(config-ephone-dn) number 5001
Router(config-ephone-dn) name MWI
Router(config-ephone-dn) mwi sip

Router(config) telephony-service
Router(config-telephony) mwi sip-server 172.30.0.5
```

Related Commands

	Description
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template (ephone-dn)	Applies a template to an ephone-dn configuration.
mwi sip-server (telephony-service)	Configures the IP address and port number for the external SIP-based MWI server.
show mwi relay clients	Displays registration information for MWI relay clients.

mwi sip-server

To configure parameters associated with an external SIP-based message-waiting indication (MWI) server, use the **mwi sip-server** command in telephony-service configuration mode. To disable MWI server functionality, use the **no** form of this command.

mwi sip-server *ip-address* [{**transport tcp** | **transport udp**}] [**port** *port-number*] [**reg-e164**] [**unsolicited** [**prefix** *prefix-string*]]
no mwi sip-server *ip-address* [{**transport tcp** | **transport udp**}] [**port** *port-number*] [**reg-e164**] [**unsolicited** [**prefix** *prefix-string*]]

Syntax Description

ip-address	IP address of the MWI server.
transport tcp	(Optional) Selects TCP as the transport layer protocol. This is the default transport protocol.
transport udp	(Optional) Selects UDP as the transport layer protocol. The default if these keywords are not used is TCP.
port port-number	(Optional) Specifies port number for the MWI server. Range is from 2000 to 9999. Default is 5060 (SIP standard port).
reg-e164	(Optional) Registers an E.164 number with a Session Interface Protocol (SIP) proxy or registrar rather than an extension number. Registering with an extension number is the default.
unsolicited	(Optional) Sends SIP Notify message for MWI without any need to send a Subscribe message from the Cisco Unified CME router.
prefix *prefix-string*	(Optional) Allows the specified digits to be present before a recognized Cisco Unified CME extension number. The maximum prefix length is 32 digits.

Command Default

An external SIP-based MWI server is not defined.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)ZJ	Cisco CME 3.0	The **unsolicited** keyword was added.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	The **prefix** *prefix-string* keyword-argument pair was added.
12.4(9)T	Cisco Unified CME 4.0	This command with the **prefix** *prefix-string* keyword-argument pair was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to configure the IP address of an external SIP MWI server. This IP address is used with the **mwi sip** (ephone-dn) command to subscribe individual ephone-dn extension numbers to the notification list of the MWI SIP server. A SIP MWI client runs TCP by default.

The **transport tcp** keyword is the default setting. The **transport udp** keyword allows you to integrate with a SIP MWI client. The optional **port** keyword is used to specify a port number other than 5060, the default. The default registration is with an extension number, so the **reg-e164** keyword allows you to register with an E.164 ten-digit number.

Central voice-messaging servers that provide mailboxes for several Cisco Unified CME sites may use site codes or prefixes to distinguish among similarly numbered ranges of extensions at different sites. In Cisco CME 3.2.3 and later versions, you can specify that your Cisco Unified CME system should accept unsolicited SIP Notify messages for message-waiting indication (MWI) that include a prefix string as a site identifier.

Examples

The following example sets MWI for the SIP server and sets individual ephone-dn extension numbers to the MWI SIP server's notification list:

```
Router(config) ephone-dn 1
Router(config-ephone-dn) number 5001
Router(config-ephone-dn) name Accounting
Router(config-ephone-dn) mwi sip
Router(config-ephone-dn) exit
Router(config) telephony-service
Router(config-telephony) mwi sip-server 192.168.0.5 transport udp
```

The following example identifies the SIP server for MWI notification at the IP address 172.16.14.22. It states that the Cisco Unified CME system will accept unsolicited SIP Notify messages that include the prefix 555 as a site identifier.

```
telephony-service
 mwi sip-server 172.16.14.22 unsolicited prefix 555
```

Related Commands

	Description
mwi (ephone-dn)	Configures specific Cisco Unified IP phone directory numbers to receive MWI notification from an external voice-mail system.
mwi expires	Sets the expiration timer for registration for the client or the server.
mwi sip (ephone-dn)	Subscribes an extension in a Cisco Unified CME router to receive MWI notification from a SIP MWI server.
show mwi relay clients	Displays the registration information for MWI relay clients.

mwi stutter (voice register global)

To generate a stutter tone for message-waiting indication (MWI) in a Cisco CallManager Express (Cisco CME) system using SIP, use the **mwi stutter** command in voice register global configuration mode. To disable MWI stutter, use the **no** form of this command.

mwi stutter
no mwi stutter

Syntax Description

This command has no arguments or keywords.

Command Default

Stutter tone for MWI is disabled.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Examples

The following example shows how to enable MWI stutter:

```
Router(config)# voice register global
Router(config-register-global)# mwi stutter
```

Related Commands

	Description
mwi reg-e164	Registers full E.164 number to the MWI server in Cisco Unified CME and enables MWI.

mwi-line

To designate a line other than the primary line of an ephone to be associated with the ephone's message waiting indicator (MWI) lamp, use the **mwi-line** command in ephone configuration mode. To return to the default, use the **no** form of this command.

mwi-line *line-number*
no mwi-line

Syntax Description

line-number	Line number to be associated with the MWI lamp. Range is from 1 to 34.

Command Default

A phone's MWI lamp is lit only when there is a message waiting for the phone's primary line (line 1).

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command designates a phone line other than the primary line to activate the MWI lamp on the phone. When a message is waiting for an ephone-dn associated with the designated line, the MWI lamp is turned on. When the message is heard, the MWI lamp is turned off. For phone lines other than the line that is designated to receive MWI, an envelope icon is displayed next to them when there is a message waiting.

Note that a logical phone "line" is not the same as a phone button. A line is a button that has one or more ephone-dns assigned to it. A button that has no ephone-dns assigned to it does not count as a line.

In most cases, one ephone-dn is assigned to one button on an ephone. When you set the **mwi-line** command to that button, the MWI lamp is turned on when there is a message waiting for that ephone-dn. When you set the **mwi-line** command to a button with a more complex configuration, the following rules apply:

- When a button has a single ephone-dn with primary and secondary numbers, the MWI lamp is turned on only when there is a message waiting for the primary number.
- When a button has several ephone-dns overlaid on it, the MWI lamp is turned on only when there is a message waiting for the first number in the list of ephone-dns.
- When a button is an overflow button for an overlay button, the MWI lamp is not turned on for any extension that might overflow to this button. If you set the **mwi-line** command to this button, the command is ignored.

Examples

The following example enables MWI on ephone 18 for line 2 (button 2), which has overlaid ephone-dns. The MWI lamp on this phone will be lit only if there is a message waiting for extension 2021. Button 4 is unused. The line numbers in this example are as follows:

- Line 1—Button 1—Extension 2020
- Line 2—Button 2—Extension 2021, 2022, 2023, 2024, 2025
- Line 3—Button 3—Extension 2021, 2022, 2023, 2024, 2025 (rollover line)
- Button 4—Unused

- Line 4—Button 5—Extension 2026

```
ephone-dn 20
 number 2020
ephone-dn 21
 number 2021
ephone-dn 22
 number 2022
ephone-dn 23
 number 2023
ephone-dn 24
 number 2024
ephone-dn 25
 number 2025
ephone-dn 26
 number 2026
ephone 18
 button 1:20 2o21,22,23,24,25 3x2 5:26
 mwi-line 2
```

The following example enables MWI on ephone 17 for line 3 (extension 609). In this case, the button numbers do not match the line numbers because buttons 2 and 4 are not used.

- Line 1—Button 1—Extension 607
- Button 2—Unused
- Line 2—Button 3—Extension 608
- Button 4—Unused
- Line 3—Button 5—Extension 609

```
ephone-dn 17
 number 607
ephone-dn 18
 number 608
ephone-dn 19
 number 609
ephone 25
 button 1:17 3:18 5:19
 mwi-line 3
```

Related Commands

	Description
button	Associates ephone-dns with individual buttons on an SCCP phone and to specify line type or ring behavior.

mwi-type

To specify the type of message-waiting indication (MWI) notification that a directory number can receive and process, use the **mwi-type** command in ephone-dn or ephone-dn-template configuration mode. To disable this feature, use the **no** form of this command.

mwi-type {**visual** | **audio** | **both**}
no mwi-type {**visual** | **audio** | **both**}

Syntax Description

visual	Sets a directory number to process visual MWI, using either the main or secondary phone number.
audio	Sets a directory number to process audible MWI (AMWI), using either the main or secondary phone number.
both	Sets a directory number to process both visual and audible MWI, using either the main or secondary phone number.

Command Default

If MWI is enabled for a directory number, directory number will receive visual MWI.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(6)XE	Cisco Unified CME 4.0(2)	This command was introduced.
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)T	Cisco Unified CME 4.0(3)	This command was integrated into Cisco IOS Release 12.4(11)T.

Usage Guidelines

This command enables a directory number to receive audible, visual, or both audible and visual MWI notification from an external voice-messaging system. The external voice-messaging system is able to communicate MWI status by making telephone calls to the dummy extension, with the MWI information embedded in either the called or calling-party IP phone number.

Based on the capabilities of the IP phone and how the **mwi-type** command is configured, Message Waiting is communicated as follows:

- If the phone supports (visual) MWI and MWI is configured for the phone, Message Waiting light is lit.
- If the phone supports (visual) MWI only, Message Waiting light is lit regardless of the configuration.
- If the phone supports AMWI and AMWI is configured for the phone, stutter dial tone is sent to the phone when it goes off-hook.
- If the phone supports AMWI only and AMWI is configured, stutter dial tone is sent to the phone when it goes off hook regardless of the configuration.
- If a phone supports (visual) MWI and AMWI and both options are configured for the phone, the Message Waiting light is lit and a stutter dial tone is sent to the phone when it goes off-hook.

Before using this command:

- Create the directory number to be configured by using the **number**

- Enable MWI on this directory number by using the **mwi** command.

If you use an ephone-dn template to apply a command to a directory number and you also use the same command in ephone-dn configuration mode for the same number, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example shows how to enable AMWI on extension 8000, assuming that the phone to which this directory number is assigned supports AMWI. Otherwise, a call placed by the voice-mail system to 8001 turns on the MWI light for the extension number indicated by the calling-party number for the MWI call.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number 8000
Router(config-ephone-dn) MWI on
Router(config-ephone-dn) MWI-type audible
```

The following example shows how to enable both audible and visual MWI. A call placed by the voice-mail system to 8001 turns on the MWI light for the extension number indicated by the calling-party number for the MWI call. When the phone user takes the phone off hook, they hear a stutter dial tone:

```
Router(config)# ephone-dn 2
Router(config-ephone-dn) number 8001
Router(config-ephone-dn) MWI on
Router(config-ephone-dn) MWI-type both
```

The following example shows how to use an ephone-dn-template to set MWI type:

```
Router(config)# ephone-dn-template 4
Router(config-ephone-dn-template) MWI-type both
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 8000
Router(config-ephone-dn)# ephone-dn-template 4
```

Related Commands

	Description
ephone-dn-template (ephone-dn)	Applies a template to an ephone-dn configuration.
mwi (ephone and ephone template)	Enables a directory number to receive MWI.
number	Associates a telephone or extension number with a directory number in a Cisco Unified CME system.

Cisco Unified CME Commands: N

name (ephone-dn)

To associate a name with a directory number in Cisco Unified CME, use the **name** command in ephone-dn configuration mode. To disassociate a name from an extension, use the **no** form of this command.

name *name*
no name

Syntax Description

name	Alphanumeric string of person or group associated with a directory number. Name must follow the order specified in the **directory (telephony-service)** command, either **first-name-first** or **last-name-first**. The two parts, first and last name or last and first name, of this argument must be separated with a space.

Command Default

This command has no default behavior or values.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

The *name* argument is used to provide caller ID for calls originating from a directory number in Cisco Unified CME and also generates local directory information that is accessed by using the Directories button on a Cisco IP phone.

The *name* argument combination must match the order specified in the **directory (telephony-service)** command, either **first-name-first** or **last-name-first**.

The *name* string must contain a space between the first and second parts of the string (first last or last first).

The *name* string cannot contain special characters such as an ampersand (&). The only special characters supported in the name string are the comma (,) and the percent sign (%).

To display a comma between the last and first names when the pattern is last-name-first, add a comma (,) to the end of the first part of the *name* string (last name), for example: last, first.

The second part of the *name* string can contain spaces, such as “and Handling.”

Examples

The following example configures the username John Smith with the pattern **first-name-first**:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) name John Smith
```

The following example configures the username Shipping and Handling with the pattern **first-name-first**:

```
Router(config)# ephone-dn 1
```

```
Router(config-ephone-dn) name Shipping and Handling
```

The following example configures the username Jane Smith with the pattern **last-name-first** and with a comma:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) name Smith, Jane
```

Related Commands

Command	Description
directory (telephony-service)	Defines the name order for the local directory of Cisco IP phone users.

name (ephone-hunt)

To associate a name with a called voice hunt group, use the **name** command in ephone-hunt configuration mode. To dissociate the name of the called voice hunt group, use the **no** form of this command.

name*“primary pilot name“***[secondary** *“secondary "secondary pilot name”***]**

no name*“primary pilot name“***[secondary** *“secondary "secondary pilot name”***]**

Syntax Description

“primary pilot name”	Name of primary pilot number.
secondary *“secondary pilot name”*	(Optional) Name of secondary pilot number.

Command Default

No name is associated with the called voice hunt group.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

In Cisco Unified CME 9.5 and Cisco Unified SRST 9.5, when the secondary pilot name is not explicitly configured, the primary pilot name is applicable to both pilot numbers.

Note Use quotes (") when input strings have spaces in between.

Examples

The following example configures the primary pilot name for both the primary and the secondary pilot numbers:

```
name SALES
```

The following example configures different names for the primary and secondary pilot numbers:

```
name SALES secondary SALES-SECONDARY
```

The following example associates a two-word name for the primary pilot number and a one-word name for the secondary pilot number:

```
name “CUSTOMER SERVICE” secondary CS
```

The following example associates a one-word name for the primary pilot number and a two-word name for the secondary pilot number:

```
name FINANCE secondary “INTERNAL ACCOUNTING”
```

The following example associates two-word names for the primary pilot number and the secondary pilot number:

name "INTERNAL CALLER" secondary "EXTERNAL CALLER"

When incoming call A reaches voice hunt group B and lands on final C, extension C does not show the name of the forwarder because the voice hunt group is not configured to display the name. To display the name of the forwarder and the final number, two separate names are required for the primary and secondary pilot numbers.

The following is a sample output of the **show run** command when the primary and secondary pilot names are configured in ephone-hunt configuration mode:

```
ephone-hunt 10 sequential
 pilot 1010 secondary 1020
 list 2004, 2005
 final 2006
 timeout 8, 8
 name "EHUNT PRIMARY" secondary "EHUNT SECONDARY"
ephone-hunt 11 peer
 pilot 1012 secondary 1022
 list 2004, 2005
 final 2006
 timeout 8, 8
 name EHUNT1 secondary EHUNT1-SEC
```

The following is a sample output of the **show ephone-hunt** command when the primary and secondary pilot names are configured in ephone-hunt configuration mode:

```
show ephone-hunt 10
Group 10
    type: sequential
    pilot number: 1010, peer-tag 20010
    pilot name: EHUNT PRIMARY
    secondary number: 1020, peer-tag 20011
```

secondary name: EHUNT SECONDARY

Related Commands

Command	Description
voice hunt-group	Enters voice hunt-group configuration mode and creates a hunt group for phones in a Cisco Unified CME system.
show voice hunt-group	Displays configuration information associated with one or all voice hunt groups in a Cisco Unified CME system.

name (voice emergency response location)

To describe or identify an emergency response location, use the **name** command in voice emergency response location mode. To remove this definition, use the **no** form of this command.

name *string*
no name

Syntax Description

string	String (30 characters) used to describe or identify an ERL's location.

Command Default

The location is not described.

Command Modes

Voice emergency response location configuration (cfg-emrgncy-resp-location)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to enable a word or description of the ERL for administrative purposes. The most common use of this command is to identify the location for the network administrator.

Examples

In this example, the location description is Your Company Incorporated.

```
voice emergency response location 60
subnet 1 209.165.200.224 255.255.0.0
elin 1 4085550101
name Your Company Incorporated,
```

Related Commands

Command	Description
address	Specifies a comma separated text entry (up to 247 characters) of an ERL's civic address.
elin	Specifies a PSTN number to replace the caller's extension.
subnet	Defines which IP phones are part of this ERL.
voice emergency response location	Creates a tag for identifying an ERL for E911 services.

name (voice hunt-group)

To associate a name with a called voice hunt group, use the **name** command in voice hunt-group configuration mode. To dissociate the name of the called voice hunt group, use the **no** form of this command.

name*“primary pilot name“***[secondary***“secondary pilot name”***]**
no name*“primary pilot name“***[secondary***“secondary pilot name”***]**

Syntax Description

“primary pilot name”	Name of primary pilot number.
secondary *“secondary pilot name”*	(Optional) Name of secondary pilot number.

Command Default

No name is associated with the called voice hunt group.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

In Cisco Unified CME 9.5 and Cisco Unified SRST 9.5, when the secondary pilot name is not explicitly configured, the primary pilot name is applicable to both pilot numbers.

Note Use quotes (") when input strings have spaces in between.

Examples

The following example configures the primary pilot name for both the primary and the secondary pilot numbers:

```
name SALES
```

The following example configures different names for the primary and secondary pilot numbers:

```
name SALES secondary SALES-SECONDARY
```

The following example associates a two-word name for the primary pilot number and a one-word name for the secondary pilot number:

```
name “CUSTOMER SERVICE” secondary CS
```

The following example associates a one-word name for the primary pilot number and a two-word name for the secondary pilot number:

```
name FINANCE secondary “INTERNAL ACCOUNTING”
```

The following example associates two-word names for the primary and secondary pilot numbers:

```
name “INTERNAL CALLER” secondary “EXTERNAL CALLER”
```

When incoming call A reaches voice hunt group B and lands on final C, extension C does not show the name of the forwarder because the voice hunt group is not configured to display the name. To display the name of the forwarder and the final number, two separate names are required for the primary and secondary pilots.

The following example shows how the primary and secondary pilot names are configured in voice hunt-group configuration mode:

```
voice hunt-group 24 parallel
   final 097
   list 885,886,124,154
   timeout 20
   pilot 021 secondary 621
   name SALES secondary SALES-SECONDARY
```

The following is a sample output of the **show voice hunt-group** command when the primary and secondary pilot names are configured in voice hunt-group configuration mode:

```
show voice hunt-group 1
Group 1
    type: parallel
    pilot number: 1000, peer-tag 2147483647
    secondary number: 2000, peer-tag 2147483646
    pilot name: SALES
    secondary name: SALES-SECONDARY
    list of numbers: 2004,2005
```

Related Commands

Command	Description
voice hunt-group	Enters voice hunt-group configuration mode and creates a hunt group for phones in a Cisco Unified CME system.
show voice hunt-group	Displays configuration information associated with one or all voice hunt groups in a Cisco Unified CME system.

name (voice register dn)

To associate a name with a directory number in Cisco Unified CME, use the **name** command in voice register dn configuration mode. To disassociate a name from an extension, use the **no** form of this command.

name *name*
no name

Syntax Description

name	Name of the person associated with a given extension. Name must follow the order specified in the **directory** (telephony-service) command, either **first-name-first** or **last-name-first**.

Command Default

This command has no default behavior or values.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

The *name* argument is used to provide caller ID for calls originating from a Cisco CME extension.

This command also generates directory information for the local directory that is accessed from the Directories button on a Cisco IP phone.

Examples

The following example shows how to configure the username John Smith with the pattern **first-name-first**:

```
Router(config)# voice register dn 1
Router(config-register-dn) name John Smith
```

The following example shows how to configure the username Jane Smith with the pattern **last-name-first**:

```
Router(config)# voice register dn 1
Router(config-register-dn) name Smith, Jane
```

Related Commands

	Description
directory (telephony-service)	Defines the name order for the local directory of Cisco IP phone users.

network-locale (ephone-template)

To specify a network locale in an ephone template, use the **network-locale** command in ephone-template configuration mode. To reset to the default network locale, use the **no** form of this command.

network-locale *network-locale-tag*
no network-locale

Syntax Description

network-locale-tag	Locale identifier that was assigned to a network locale using the **network-locale (telephony-service)** command.

Command Default

The default network locale (network locale 0) is used.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

To apply network locales to individual ephones, you must specify per-phone configuration files using the **cnf-file perphone** command and identify the locales using the **network-locale (telephony-service) command.**

After creating an ephone template that contains a locale tag, use the **ephone-template (ephone)** command to apply the template to individual ephones.

Examples

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 create cnf-files
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
ephone-template 1
 user-locale 1
 network-locale 1
ephone-template 2
 user-locale 2
 network-locale 2
ephone-template 3
 user-locale 3
 network-locale 3
ephone 11
```

```
 button 1:25
 ephone-template 1
ephone 12
 button 1:26
 ephone-template 2
ephone 13
 button 1:27
 ephone-template 3
ephone 14
 button 1:28
```

Related Commands

	Description
cnf-file	Specifies the type of configuration files that phones use.
ephone-template (ephone)	Applies an ephone template to an ephone.
network-locale (telephony-service)	Sets the locale for geographically specific tones and cadences.

network-locale (telephony-service)

To select a code for a geographically specific set of tones and cadences on supported phone types, use the **network-locale** command in telephony-service configuration mode. To disable selection of a code, use the **no** form of this command.

network-locale [**network-locale-tag** [**user-defined-code**]] **locale-code**
no network-locale network-locale-tag

Syntax Description

network-locale-tag	(Optional) Assigns a locale identifier to the locale code. Range is 0 to 4. Default is 0.
user-defined code	(Optional) Assigns one of the user-defined codes to the specified locale code. Valid codes are **U1**, **U2**, **U3**, **U4**, and **U5**. There is no default.
locale-code	Locale files for the following ISO 3166 codes are predefined in system storage for supported phone types: • **AT**—Austria • **CA**—Canada • **CH**—Switzerland • **DE**—Germany • **DK**—Denmark • **ES**—Spain • **FR**—France • **GB**—United Kingdom • **IT**—Italy • **JP**—Japan • **NL**—Netherlands • **NO**—Norway • **PT**—Portugal • **RU**—Russian Federation • **SE**—Sweden • **US**—United States (default) **Note** You can also assign any valid ISO 3166 code that is not listed above to a user-defined code (U1 through U5), but you must first copy the appropriate XML tone files to flash, slot 0, or an external TFTP server and use the **cnf-files perphone** command to specify the use of per-phone configuration files.

Command Default

The default locale code is **US** (United States).

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.

Cisco IOS Release	Cisco Product	Modification
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.4(4)XC	Cisco Unified CME 4.0	The *network-locale-tag* and *user-defined-code* arguments were added.
12.4(9)T	Cisco Unified CME 4.0	The *network-locale-tag* and *user-defined-code* arguments were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command designates a a network locale other than US as the locale for one or more phones in Cisco Unified CME.

Network locale 0 always holds the default locale, which is used for all phones that are not assigned alternative network locales or user-defined network locales. You can use this command to change the default locale.

The **show telephony-service tftp-bindings** command displays the locale-specific call-progress tone files that are accessible to IP phones using TFTP.

This command must be followed by a complete phone reboot using the **reset** command.

Alternative Network Locales

The *network-locale-tag* argument allows you to specify up to five alternative network locales for use in a system using Cisco Unified CME 4.0 or a later release. For example, a company can specify network-locale France for phones A, B, and C; network-locale Germany for phones D, E, and F; and network-locale United States for phones G, H, and I.

Each one of the five alternative network locales that you can use in a multi-locale system is identified with a locale tag identifier. The identifier 0 always holds the default locale, although you can define this default to be any locale code that is supported in the system and is listed in the CLI help for the command. For example, if you define network locale 0 to be JP (Japanese), the default network locale for the router is JP. If you do not specify a locale for the identifier 0, the default is US (United States).

To apply alternative network locales to different phones, you must use the **cnf-files** command to specify per-phone configuration files. When you use per-phone configuration files, a phone's configuration file automatically uses the default locales in user locale 0 and network locale 0. You can override this default for individual ephones by assigning alternative locale tag identifiers to the alternative locale codes that you want to use and then creating ephone templates to assign the locale tag identifiers to individual ephones. For example, you can give the alternative locale tag of 2 to the locale code DK (Denmark).

After using the **network-locale (telephony-service)** command to associate a locale tag identifier with a locale code, use the **network-locale** command in ephone-template mode to apply the locale tag to an ephone template. Then use the **ephone-template** command in ephone configuration mode to apply the template to the ephones that should use the alternative network locale.

User-Defined Network Locales

XML files for user locales and network locales that are not currently provided in the system must be downloaded to use this feature. Beginning in Cisco Unified CME 4.0, you can install the files to support a particular user and network locale in flash, slot 0, or an external TFTP server. You cannot install these files in the system location. These user-locale and network-locale files can then be used as default or alternative locales for all or some phones.

For example, if you have a site at which the phones should use the displays and tones for Traditional Chinese, which is not one of the predefined locales, you must download and install the XML files for Traditional Chinese on the phones that need to use this locale.

Examples

The following example sets the default locale tag 0 to France:

```
telephony-service
 network-locale FR
```

The following example sets the default locale tag 0 to France. It shows another way to change the default network locale:

```
telephony-service
 network-locale 0 FR
```

The following example sets the alternative locale tag 1 to Germany:

```
telephony-service
 network-locale 1 DE
```

Alternative Network Locale Example

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 create cnf-files
ephone-template 1
 user-locale 1
 network-locale 1
ephone-template 2
 user-locale 2
 network-locale 2
ephone-template 3
 user-locale 3
 network-locale 3
ephone 11
 button 1:25
 ephone-template 1
ephone 12
 button 1:26
 ephone-template 2
ephone 13
 button 1:27
 ephone-template 3
ephone 14
```

```
 button 1:28
```

User-Defined Network Locale Example

The following example applies the alternative locale tag 4 to the user-defined code U1, which is defined as ZH. ZH is the code that represents Traditional Chinese in ISO 639, the Language Code Reference. Because the code for Traditional Chinese is not one of those is provided in the system, the user must download the appropriate XML files to support this language.

In addition to the user-defined code, the example also defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales; ephone 12 uses FR; ephone 13 uses ES; ephone 14 uses the default, US; and ephone 15 uses the user-defined language, ZH (Traditional Chinese).

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 user-locale 4 U1 ZH
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 network-locale 4 U1 ZH
 create cnf-files
ephone-template 1
 user-locale 1
 network-locale 1
ephone-template 2
 user-locale 2
 network-locale 2
ephone-template 3
 user-locale 3
 network-locale 3
ephone-template 4
 user-locale 4
 network-locale 4
ephone 11
 button 1:25
 ephone-template 1
ephone 12
 button 1:26
 ephone-template 2
ephone 13
 button 1:27
 ephone-template 3
ephone 14
 button 1:28
ephone 15
 button 1:29
 ephone-template 4
```

Related Commands

	Description
cnf-files	Specifies the type of phone configuration files to be created.
ephone-template (ephone)	Applies an ephone template to an ephone.
network-locale (ephone-template)	Applies a locale tag identifier to an ephone template.
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service tftp-bindings	Displays the current configuration files that are accessible to IP phones.
user-locale (telephony-service)	Sets the language for displays on supported phone types.

network-locale (voice-gateway)

To select a geographically specific set of tones and cadences for the voice gateway's analog endpoints that register to Cisco Unified CME, use the **network-locale** command in voice-gateway configuration mode. To remove a code, use the **no** form of this command.

network-locale *country-code*
no network-locale *country-code*

Syntax Description

country-code	The following ISO 3166 country codes are supported: • **AE**—United Arab Emirates • **AR**—Argentina • **AT**—Austria • **AU**—Australia • **BE**—Belgium • **BR** —Brazil • **CA**—Canada • **CH**—Switzerland • **CN**—China • **CO**—Colombia • **CY**—Cyprus • **CZ**—Czech Republic • **DE**—Germany • **DK**—Denmark • **EG**—Egypt • **ES**—Spain • **FI**—Finland • **FR**—France • **GB**—United Kingdom • **GH**—Ghana • **GR**—Greece • **HK**—Hong Kong • **HU**—Hungary • **ID**—Indonesia • **IE**—Ireland • **IL**—Israel • **IN**—India • **IS**—Iceland • **IT**—Italy • **JO**—Jordan • **JP**—Japan • **KE**—Kenya • **KR**—Korea Republic • **KW**—Kuwait • **LB**—Lebanon • **LU**—Luxembourg • **MX**—Mexico • **MY**—Malaysia • **NG**—Nigeria • **NL**—Netherlands • **NO**—Norway • **NP**—Nepal • **NZ**—New Zealand • **OM**—Oman • **PA**—Panama • **PE**—Peru • **PH**—Philippines • **PK**—Pakistan • **PL**—Poland • **PT**—Portugal • **RU**—Russian Federation • **SA**—Saudi Arabia • **SE**—Sweden • **SG**—Singapore • **SI**—Slovenia • **SK**—Slovakia • **TH**—Thailand • **TR**—Turkey • **TW**—Taiwan • **US**—United States (default) • **VE**—Venezuela • **ZA**—South Africa • **ZW**—Zimbabwe

Command Default

The default locale code is **US** (United States).

Command Modes

Voice-gateway configuration (config-voice-gateway)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.

Cisco IOS Release	Cisco Product	Modification
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command designates a network locale other than US as the locale for the analog endpoints registered to Cisco Unified CME. All voice ports are assigned the same network locale. If you want a different network locale on a specific phone, use the **cptone** command in voice-port configuration mode.

The **show telephony-service tftp-bindings** command displays the locale-specific call-progress tone files that are accessible to IP phones using TFTP.

After using this command, you must reboot the phones with the **reset** command.

Examples

The following example shows a voice gateway configuration where the network locale is set to France:

```
voice-gateway system  1
 network-locale FR
 type VG224
 mac-address 001F.A30F.8331
 voice-port 0-23
 create cnf-files
```

Related Commands

Command	Description
cptone	Specifies a regional analog voice-interface-related tone, ring, and cadence setting.
reset (voice-gateway)	Performs a complete reboot of all analog phones associated with the voice gateway and registered to Cisco Unified CME.
show telephony-service tftp-bindings	Displays the current configuration files accessible to IP phones.
voice-port (voice-gateway)	Identifies the ports on the voice gateway that will register to Cisco Unified CME.

night-service bell

To mark an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods, use the **night-service bell** command in ephone or ephone-template configuration mode. To remove night-service notification capability from a phone, use the **no** form of this command.

night-service bell
no night-service bell

Syntax Description

This command has no arguments or keywords.

Command Default

A phone is not marked for night-service bell notification.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

When an ephone-dn is marked for night-service treatment using the **night-service bell** (ephone-dn) command, incoming calls that ring during the night-service time period on that ephone-dn send an alert indication to all IP phones that are marked to receive night-service bell notification with this command. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the ephone-dn extension number. The phone user retrieves the call by pressing a PickUp or GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the **night-service date** and **night-service day** commands. Night service can be manually disabled or reenabled from a phone configured with ephone-dns in night-service mode if the **night-service code** command has been set.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example designates the IP phone that is being configured as a phone that will receive night-service bell notification when ephone-dns marked for night service receive incoming calls during a night-service period:

```
Router(config)# ephone 4
Router(config-ephone)# night-service bell
```

Related Commands

	Description
ephone-template (ephone)	Applies a template to an ephone configuration.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

night-service bell (ephone-dn)

To mark an ephone-dn for night-service treatment, use the **night-service bell** command in ephone-dn configuration mode. To remove the night-service treatment from the ephone-dn, use the **no** form of this command.

night-service bell
no night-service bell

Syntax Description

This command has no arguments or keywords.

Command Default

An ephone-dn is not marked for night service.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

When an ephone-dn is marked for night-service treatment using this command, incoming calls that ring during the night-service time period on that ephone-dn send an alert indication to all IP phones that are marked to receive night-service bell notification using the **night-service bell (ephone) command.** The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the ephone-dn extension number. The phone user retrieves the call by pressing a PickUp or GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the **night-service date** and **night-service day** commands. Night service can be manually disabled or enabled from a phone configured with ephone-dns in night-service mode if the **night-service code** command has been set.

Examples

The following example marks an ephone-dn as a line that will ring on IP phones designated to receive night-service bell notification when incoming calls are received on this ephone-dn during night-service periods:

```
Router(config)# ephone-dn 16
Router(config-ephone-dn)# night-service bell
```

Related Commands

	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.

	Description
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

night-service code

To define a code to disable or reenable night service on IP phones, use the **night-service code** command in telephony-service configuration mode. To remove the code, use the **no** form of this command.

night-service code *digit-string*
no night-service code *digit-string*

Syntax Description

digit-string	Digit code that a user enters at an IP phone to disable or reenable night service. The code must begin with an asterisk (*). The maximum number of characters is 16, including the asterisk.

Command Default

No code is defined.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(14)T	Cisco CME 3.3	The action of this command was changed so that all night-service ephone-dns are activated or deactivated when the code is used rather than just the phone on which the code is input.
15.6(3)M 16.3.1	Cisco Unified CME 11.5	From Unified CME 11.5 onwards, this command can be used to activate or deactivate the night service from SIP phones as well.

Usage Guidelines

Night-service periods are defined with the **night-service date** and **night-service day** commands. When a dn is marked for night-service treatment using the **night-service bell** (ephone-dn or voice register dn) command, incoming calls that ring during the night-service time period on that dn send an alert indication to all IP phones that are marked to receive night-service bell notification using the **night-service bell** command. The alert notification is in the form of a burst ring for SCCP phones and message waiting tone for SIP phones (not associated with any of the individual lines on the IP phone). There is a visible display of the dn extension number. The phone user retrieves the call by pressing a PickUp or GPickUp soft key and dialing the appropriate digits.

When a night-service code has been defined using the **night-service code** command, night service for all night-service dns can be manually activated or deactivated from any phone that is configured with a night-service dn.

Examples

The following example defines a night-service code of *2985:

```
Router(config)# telephony-service
Router(config-telephony)# night-service code *2985
```

Related Commands

	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service bell (voice register pool)	Marks a SIP phone to receive night-service bell notification when incoming calls are received on voice register DNs that are marked for night service during night-service time periods.
night-service bell (voice register dn)	Marks a voice register dn to send night-service bell notification to designated SIP phones during night-service time periods.
night-service bell (voice register template)	Applies a template to a pool configuration.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

night-service date

To define a recurring time period associated with a date during which night service is active, use the **night-service date** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service date *month day start-time stop-time*
no night-service date *month day start-time stop-time*

Syntax Description

month	Abbreviated month. The following abbreviations for month are valid: **jan**, **feb**, **mar**, **apr**, **may**, **jun**, **jul**, **aug**, **sep**, **oct**, **nov**, **dec**.
day	Day of the month. Range is from 1 to 31.
start-time stop-time	Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. The stop time must be greater than the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified date.

Command Default

No time period based on date is defined for night service.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

After you define night-service periods using this command and the **night-service day** command, use the **night-service bell** (ephone-dn) command to specify the extensions that will ring on other phones and the **night-service bell** (ephone) command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period for the entire day of January 1:

```
Router(config)# telephony-service
Router(config-telephony)# night-service date jan 1 00:00 00:00
```

Related Commands

	Description
night-service bell (ephone)	Marks an IP phone to receive night-service-bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.

	Description
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

night-service day

To define a recurring time period associated with a day of the week during which night service is active, use the **night-service day** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service day *day start-time stop-time*
no night-service day *day start-time stop-time*

Syntax Description

day	Day of the week abbreviation. The following are valid day abbreviations: **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, **sat**.
start-time *stop-time*	Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means "from Monday at 7 p.m. until Tuesday at 7 a.m." The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

Command Default

No time period based on day of the week is defined for night service.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

After you define night-service periods using this command and the **night-service date** command, use the **night-service bell (ephone-dn)** command to specify the extensions that will ring on other phones and the **night-service bell (ephone)** command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period from Monday at 7 p.m. to Tuesday at 9 a.m.:

```
Router(config)# telephony-service
Router(config-telephony)# night-service day mon 19:00 09:00
```

Related Commands

	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.

	Description
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.

night-service everyday

To define a recurring time period during which night service is active every day, use the **night-service everyday** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service everyday *start-time stop-time*
no night-service everyday

Syntax Description

start-time *stop-time*	Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means "from Monday at 7 p.m. until Tuesday at 7 a.m." The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

Command Default

No recurring night-service time period is defined for every day.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

After you define recurring night-service time periods, use the **night-service bell (ephone-dn)** command to specify the extensions that will ring on other phones and the **night-service bell (ephone)** command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period to be in effect every day from 7 p.m. to 8 a.m.:

```
Router(config)# telephony-service
Router(config-telephony)# night-service everyday 19:00 08:00
```

Related Commands

	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.

	Description
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.
night-service weekday	Defines a recurring night-service time period to be in effect only on weekdays.
night-service weekend	Defines a recurring night-service time period to be in effect only on weekends.

night-service weekday

To define a recurring night-service time period to be in effect on all weekdays, use the **night-service weekday** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service weekday *start-time stop-time*
no night-service weekday

Syntax Description

start-time stop-time	Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means "from Monday at 7 p.m. until Tuesday at 7 a.m." The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

Command Default

No recurring night-service time period is defined for weekdays.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Weekdays are defined as Monday, Tuesday, Wednesday, Thursday, and Friday.

After you define night-service periods, use the **night-service bell** (ephone-dn) command to specify the extensions that will ring on other phones and the **night-service bell** (ephone) command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period every weekday from 5 p.m. to 9 a.m.:

```
Router(config)# telephony-service
Router(config-telephony)# night-service weekday 17:00 09:00
```

Related Commands

	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.

	Description
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.
night-service everyday	Defines a recurring night-service time period to be in effect everyday.
night-service weekend	Defines a recurring night-service time period to be in effect only on weekends.

night-service weekend

To define a recurring night-service time period to be active on weekends, use the **night-service weekend** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service weekend *start-time stop-time*
no night-service weekend

Syntax Description

start-time *stop-time*	Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means "from Monday at 7 p.m. until Tuesday at 7 a.m." The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

Command Default

No recurring night-service time period is defined for weekends.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Weekend is defined as Saturday and Sunday.

After you define night-service periods, use the **night-service bell** (ephone-dn) command to specify the extensions that will ring on other phones and the **night-service bell** (ephone) command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period for all day Saturdays and Sundays:

```
Router(config)# telephony-service
Router(config-telephony)# night-service weekend 00:00 00:00
```

Related Commands

	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.

	Description
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.
night-service everyday	Defines a recurring night-service time period to be in effect everyday.
night-service weekday	Defines a recurring night-service time period to be in effect only on weekdays.

no-reg

To specify that the pilot number for a Cisco CallManager Express (Cisco CME) peer ephone hunt group not register with an H.323 gatekeeper, use the **no-reg** command in ephone-hunt configuration mode. To return to the default of the pilot number registering with an H.323 gatekeeper, use the **no** form of this command.

no-reg [{**both** | **pilot**}]
no no-reg [{**both** | **pilot**}]

Syntax Description

both	(Optional) Both the primary and secondary pilot numbers are not registered.
pilot	(Optional) Only the primary pilot number is not registered.

Command Default

The pilot number registers with the H.323 gatekeeper.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **both** and **pilot** keywords were introduced.

Usage Guidelines

This command is valid only for Cisco CME peer ephone hunt groups.

Examples

The following example defines peer ephone hunt group 2 with a primary and secondary pilot number, and specifies that the secondary pilot number should not register with the H.323 gatekeeper:

```
Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 2222 secondary 4444
Router(config-ephone-hunt)# no-reg
```

Related Commands

	Description
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco CME system.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.

	Description
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

no-reg (voice register dn)

To specify that a voice DN for a SIP phone line in a Cisco CallManager Express (Cisco CME) system not register with an external proxy server, use the **no-reg** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

no-reg
no no-reg

Syntax Description

This command has no arguments or keywords.

Command Default

This command is disabled.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command specifies that a particular voice DN not register with the external proxy server. Configure the **no-reg** command per line. The default is to register all SIP lines in the Cisco CME system.

Examples

The following example shows how to configure bulk registration for registering a block of phone numbers starting with 408555 with an external registrar and specify that directory number 1, number 4085550100 not register with the external registrar:

```
Router(config)# voice register global
Router(voice-register-global)# mode cme
Router(voice-register-global)# bulk 408555....
Router(voice-register-global)# exit
Router(config)# voice register dn 1
Router(config-register-dn)# number 408550100
Router(config-register-dn)# no-reg
```

Related Commands

	Description
number (voice register dn)	Associates a telephone or extension number with a SIP phone in a Cisco CME system.

nte-end-digit-delay

To specify the amount of time that each digit in the RTP NTE end event in an RFC 2833 packet is delayed before being sent, use the **nte-end-digit-delay** command in ephone or ephone-template configuration mode. To remove the delay amount, use the **no** form of this command.

nte-end-digit-delay [*milliseconds*]
no nte-end-digit-delay

Syntax Description

milliseconds	Length of delay. Range: 10 to 200. Default: 200 ms.

Command Default

All digits in the RTP NTE end event are sent in a single burst.

Command Modes

Ephone configuration (config-ephone)

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

If your system is configured for RFC 2833 DTMF interworking and if the remote system cannot handle the RTP NTE end event sent in a single burst, use this command to delay sending each digit in the RTP NTE end event by the specified number of milliseconds. The default value for the delay is 100 ms.

This command only specifies the amount of time that each digit in the RTP NTE end event is delayed before being sent. To enable the delay, you must also configure the **dtmf-interworking rtp-nte** command in voice-service or dial-peer configuration mode.

If the phone user dials digits faster than the configured RTP NTE end-event delay, Cisco Unified CME will process the dialed digits and ignore the configured RTP NTE end-event delay unless you also configure the **keypad-normalize** command in ephone or ephone-template configuration mode.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example shows the configuration for ephone 43 in which the **nte-end-digit-delay** command is configured for a 200 ms delay.

```
Router(config)# show running-config
.
.
.
ephone 43
 button 1:29
 nte-end-digit-delay 200
 keypad-normalize
```

Related Commands

Command	Description
dtmf-interworking rtp-nte	Introduces a delay between the dtmf-digit begin and dtmf-digit end events in RFC 2833 packets sent from the router.
ephone-template (ephone)	Applies a template to ephone being configured.
keypad-normalize	Ensures that the delay configured for a dtmf-end event is always honored.

ntp-server

To specify the IP address of the Network Time Protocol (NTP) server used by SIP phones in a Cisco Unified CME system, use the **ntp-server** command in voice register global configuration mode. To remove the NTP server, use the **no** form of this command.

ntp-server *ip-address* [**mode** {**anycast** | **directedbroadcast** | **multicast** | **unicast**}]
no ntp-server

Syntax Description

ip-address	IP address of the NTP server.
mode	(Optional) Enables the broadcast mode for the server.
anycast	Enables anycast mode.
directedbroadcast	Enables directed broadcast mode.
multicast	Enables multicast mode.
unicast	Enables unicast mode.

Command Default

An NTP server is not used.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command synchronizes all SIP phones to the specified NTP server.

This command is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

Examples

The following example shows the mode for the NTP server set to multicast:

```
Router(config)# voice register global
Router(config-register-global)# ntp-server 10.10.10.1 mode multicast
```

Related Commands

	Description
create profile	Generates the configuration profile files required for SIP phones.
restart (voice register)	Performs a fast reset of one or all SIP phones associated with a Cisco Unified CME router.

number (ephone-dn)

To associate a telephone or extension number with an ephone-dn in a Cisco CallManager Express (Cisco CME) system, use the **number** command in ephone-dn configuration mode. To disassociate a number from an ephone-dn, use the **no** form of this command.

number *number* [**secondary** *number*] [**no-reg** [{**both** | **primary**}]]
no number

Syntax Description

number	String of up to 16 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters when the number is dialed only by the router, as with an intercom number. One or more periods (.) can be used as wildcard characters. For details, see "Usage Guidelines."
secondary	(Optional) Associates the number that follows as an additional number for this ephone-dn.
no-reg	(Optional) The E.164 numbers in the dial peer do not register with the gatekeeper. If you do not specify an option (**both** or **primary**) after the **no-reg** keyword, only the secondary number is not registered.
both	(Optional) Both primary and secondary numbers are not registered.
primary	(Optional) Primary number is not registered.

Command Default

No primary or secondary phone number is associated with the ephone-dn.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)ZJ	Cisco CME 3.0	The ability to use alphabetic characters as part of the number string was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command defines a valid number for an ephone-dn (extension) that is to be assigned to an IP phone. The **secondary** keyword allows you to associate a second telephone number with an ephone-dn so that it can be called by dialing either the main or secondary phone number. The secondary number may contain wildcards; for example, 50.. (the number 50 followed by periods, which stand for wildcards).

The **no-reg** keyword causes an E.164 number in the dial peer not to register with the gatekeeper. If you do not specify **both** or **primary** after the **no-reg** keyword, only the secondary number does not register.

A number normally contains only numeric characters, which allow it to be dialed from any telephone keypad. However, in certain cases, such as intercom numbers, which are normally dialed only by the router, you can insert alphabetic characters into the number to prevent phone users from dialing it and using the intercom function without authorization.

A number can also contain one or more periods (.) as wildcard characters that will match any dialed number in that position. For example, 51.. rings when 5100 is dialed, when 5101 is dialed, and so forth.

After you use the **number** command, assign the ephone-dn to an ephone using the **button** command. Following the use of the **button** command, the **restart** command must be used to initiate a quick reboot of the phone to which this number is assigned.

Examples

The following example sets 5001 as the primary extension number for a Cisco IP phone and 0 as the secondary number. This configuration allows the telephone number 5001 to act as a regular extension number and also to act as the operator line such that callers who dial 0 are routed to the phone line with extension number 5001.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001 secondary 0
```

The following example sets 5001 as the primary extension number for a Cisco IP phone and "500." (the number 500 followed by a period) as the secondary number. This configuration allows any calls to extension numbers from the range 5000 to 5009 to be routed to extension 5001 if the actual extension number dialed cannot be found. For example, IP phones may be active in the system with lines that correspond to 5001, 5002, 5004, 5005, and 5009. A call to 5003 would be unable to locate a phone with extension 5003, so the call would be routed to extension 5001.

```
Router(config-ephone-dn)# number 5001 secondary 500.
```

The following example defines a pair of intercom ephone-dns that are programmed to call each other. The intercom numbers contain alphabetic characters to prevent anyone from dialing them from another phone. Ephone-dn 19 is assigned the number A5511 and is programmed to dial A5522, which belongs to ephone-dn 20. Ephone-dn 20 is programmed to dial A5511. No one else can dial these numbers.

```
Router(config)# ephone-dn 19
Router(config-ephone-dn)# number A5511
Router(config-ephone-dn)# name Intercom
Router(config-ephone-dn)# intercom A5522
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 20
Router(config-ephone-dn)# number A5522
Router(config-ephone-dn)# name Intercom
Router(config-ephone-dn)# intercom A5511
```

Related Commands

	Description
button	Associates ephone-dns with individual buttons on Cisco IP phones and specifies ring behavior per button.
intercom	Creates an intercom by programming a pair of extensions (ephone-dns) to automatically call and answer each other.
name	Configures a username associated with a directory number.
preference	Sets preference for the attached dial peer for a directory number.

	Description
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

night-service bell (voice register dn)

To mark a voice register dn for night-service treatment, use the **night-service bell** command in voice register dn configuration mode. To remove the night-service treatment from the voice register dn, use the **no** form of this command.

night-service bell
no night-service bell

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this command is disabled.

Command Modes

voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

When a voice register dn is marked for night-service treatment using this command, incoming calls that ring during the night-service time period on that voice register dn sends an alert indication to all IP phones that are marked to receive night-service bell notification. This is achieved using the **night-service bell (voice register pool)** command. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the voice register extension number. The phone user retrieves the call by pressing a GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the **night-service date** and **night-service day** commands. Night service can be manually disabled or enabled from a phone configured with voice register dn in night-service mode if the **night-service code** command has been set.

Examples

The following example marks a voice register dn as a line that will ring on IP phones designated to receive night-service bell notification when incoming calls are received on this voice register dn during night-service periods:

```
Router(config)# voice register dn 16
Router(config-register-dn)# night-service bell
```

Related Commands

Command	Description
night-service bell (voice register pool)	Marks a SIP phone to receive night-service bell notification when incoming calls are received on voice register DNs that are marked for night service during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.

Command	Description
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

night-service bell (voice register pool)

To mark a SIP phone to receive night-service bell notification when incoming calls are received on voice register dn that are marked for night service during night-service time periods, use the **night-service bell** command in voice register pool configuration mode. To remove night-service notification capability from a phone, use the **no** form of this command.

night-service bell
no night-service bell

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this command is disabled.

Command Modes

voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

When a voice register dn is marked for night-service treatment using the **night-service bell (voice register dn)** command, incoming calls that ring during the night-service time period on that DN send an alert indication to all SIP phones marked to receive night-service bell notification. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the voice register dn extension number. The phone user retrieves the call by pressing the GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the **night-service date** and **night-service day** commands. Night service can be manually disabled or re-enabled from a phone configured with voice register dn in night-service mode if the **night-service code** command is set.

If you use a voice register template to apply a command to a SIP phone and you also use the same command in pool configuration mode for the same phone, the value that you set in pool configuration mode has priority.

Examples

The following example designates the SIP phone that is being configured as a phone that will receive night-service bell notification when voice register dns marked for night service receive incoming calls during a night-service period:

```
Router(config)# voice register pool 4
Router(config-register-pool)# night-service bell
```

Related Commands

Command	Description
night-service bell (voice register dn)	Marks a voice register dn to send night-service bell notification to designated SIP phones during night-service time periods.

Command	Description
night-service bell (voice register template)	Applies a template to a pool configuration.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

night-service bell (voice register template)

To mark a SIP phone to receive night-service bell notification when incoming calls are received on voice register dn that are marked for night service during night-service time periods, use the **night-service bell** command in voice register template configuration mode. To remove night-service notification capability from a phone, use the **no** form of this command.

night-service bell
no night-service bell

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this command is disabled.

Command Modes

voice register template configuration (config-register-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

When a voice register dn is marked for night-service treatment using the **night-service bell (voice register dn)** command, incoming calls that ring during the night-service time period on that DN send an alert indication to all SIP phones marked to receive night-service bell notification. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the voice register dn extension number. The phone user retrieves the call by pressing the GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the **night-service date** and **night-service day** commands. Night service can be manually disabled or re-enabled from a phone configured with voice register dn in night-service mode if the **night-service code** command is set.

If you use a voice register template to apply a command to a SIP phone and you also use the same command in pool configuration mode for the same phone, the value that you set in pool configuration mode has priority.

Examples

The following example designates the SIP phone that is being configured as a phone that will receive night-service bell notification when voice register dns marked for night service receive incoming calls during a night-service period:

```
Router(config)# voice register pool 4
Router(config-register-pool)# night-service bell
```

Related Commands

Command	Description
night-service bell (voice register dn)	Marks a voice register dn to send night-service bell notification to designated SIP phones during night-service time periods.
voice register pool	Applies a template to a pool configuration.

Command	Description
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

number (voice register dn)

To associate a telephone or extension number with a SIP phone in a Cisco CallManager Express (Cisco CME) system, use the **number** command in voice register dn configuration mode. To disassociate a number, use the **no** form of this command.

number *number*
no number

Syntax Description

number	String of up to 16 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters when the number is dialed only by the router, as with an intercom number.

Command Default

This command has no default behavior or values.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

Valid characters in voice register DN include 0-9, '.' , '+', '*' and '#'.

To allow insertion of '#' at any place in voice register dn, the CLI "allow-hash-in-dn" is configured in voice register global mode.

When the CLI "allow-hash-in-dn" is configured, the user is required to change the dial-peer terminator from '#' (default terminator) to another valid terminator in configuration mode. The other terminators that are supported include '0'-'9', 'A'-'F', and '*'.

This command defines a valid number for an extension that is to be assigned to a SIP phone. Use this command before using the other commands in voice register dn configuration mode.

A number normally contains only numeric characters which allows users to dial the number from any telephone keypad. However, in certain cases, such as the numbers for intercom extensions, you want to use numbers that can only be dialed internally from the Cisco CallManager Express router and not from telephone keypads.

The **number** command allows you to assign alphabetic characters to the number so that the extension can be dialed by the router for intercom calls but not by unauthorized individuals from other phones.

After you use the **number** command, use the **reset** command to initiate a quick reboot of the phone to which this number is assigned.

Note This command can also be used for Cisco SIP SRST.

Examples

The following example shows how to set 5001 as the extension number for directory number 1 on a SIP phone.

```
Router(config)# voice register dn 1
Router(config-register-dn)# number 5001
```

Related Commands

	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
reset (voice register pool)	Performs a complete reboot of a single SIP phone associated with a Cisco CME router.

number (voice register pool)

To indicate the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone, use the **number** command in voice register pool configuration mode. To disable number registration, use the **no** form of this command.

number *tag* {*number-pattern* [**preference** *value*] [**huntstop**] | **dn** *dn-tag*}
no number *tag*

Syntax Description

tag	Telephone number when there are multiple **number** commands. Range is 1 to 114.
number-pattern	Phone numbers (including wild cards and patterns) that are permitted by the registrar to handle the Register message from the Cisco Unified SIP IP phone.
preference *value*	(Optional) Defines the number list preference order. Range is 0 to 10. The highest preference is 0. There is no default.
huntstop	(Optional) Stops hunting when the dial peer is busy.
dn *dn-tag*	Identifies the directory number tag for this phone number as defined by the **voice register dn** command. Range is 1 to 288.

Command Default

Cisco Unified SIP IP phones cannot register in Cisco Unified CME.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME and the **dn** keyword was added.
12.4(11)XJ	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	This command was modified. The *number-pattern* argument and **preference** and **huntstop** keywords were removed from Cisco Unified CME.
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	The modifications to this command were integrated into Cisco IOS Release 12.4(15)T.
15.2(4)M	Cisco Unified CME 9.1 Cisco Unified SIP SRST 9.1	This command was modified to increase the valid value of the *tag* argument to 114.

Usage Guidelines

The **number** command indicates the phone numbers that are permitted by the registrar to handle the Register message from the Cisco Unified SIP IP phone.

In Cisco Unified SRST, the keywords and arguments of this command allow for more explicit setting of user preferences regarding what number patterns should match the voice register pool.

> **Note** Configure the **id (voice register pool)** command before any other voice register pool commands, including the **number** command. The **id** command identifies a locally available, individual Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phones.

Examples

The following example shows three directory numbers assigned to Cisco Unified SIP IP phone 1 in Cisco Unified CME:

```
!
voice register pool  1
 id mac 0017.E033.0284
 type 7961
 number 1 dn 10
 number 2 dn 12
 number 3 dn 13
 codec g711ulaw
!
```

The following example shows directory numbers 10, 12, and 13 assigned to phone numbers 1, 2, and 55 of Cisco Unified SIP IP phone 2:

```
voice register pool 2
id mac 0017.E033.0284
type 7961
number 1 dn 10
number 2 dn 12
number 55 dn 13
codec g711ulaw
```

The following example shows a telephone number pattern set to 95... in Cisco Unified SRST. This means all five-digit numbers beginning with 95 are permitted by the registrar to handle the Register message.

```
voice register pool 3
 id network 10.2.161.0 mask 255.255.255.0
 number 1 95... preference 1
 cor incoming call95 1 95011
```

Related Commands

Command	Description
id (voice register pool)	Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.
voice register dn	Enters voice register dn configuration mode to define an extension for a phone line, intercom line, voice-mail port, or a message-waiting indicator.

number (voice user-profile and voice logout-profile)

To create line definitions in a voice-user profile or voice-logout profile to be downloaded to a Cisco Unified IP phone that is enabled for extension mobility, use the **number** command in voice user-profile configuration mode or voice logout-profile configuration mode. To remove line definition from a profile, use the no form of this command.

number*number* [*,...number*] type*type*

no number*number* [*,...number*] type*type*

Syntax Description

number	String of up to 16 characters that represents an E.164 telephone number to be associated with and displayed next to a line button on an IP phone. This directory number must be already configured by using the **number** command in ephone-dn or voice register dn configuration mode.
[*,...number*]	(Optional) For overlay lines only, with or without call waiting. Directory numbers to roll over to this line. Can contain up to 25 individual numbers separated by commas (,). This directory number must be already configured by using the **number** command in ephone-dn or voice register dn configuration mode.
type	Characteristics to be associated with this line button.
type	Word that describes characteristics to be associated with the line button being configured. Valid entries are as follows: • beep-ring • feature-ring • monitor-ring • silent-ring • overlay • cw-overlay

Command Default

No line definition is created.

Command Modes

Voice logout-profile configuration (config-logout-profile)
Voice user-profile configuration (config-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command in voice user-profile configuration mode to creates a line button definition in a user profile to be downloaded to the IP phone when the user is logged into an IP phone that is enabled for extension mobility.

This command in voice logout-profile configuration mode creates a line button definition in a default profile to be downloaded to an IP phone when no user is logged into an IP phone that is enabled for extension mobility.

For button appearance, extension mobility will associate line definitions in the voice-logout profile or voice-user profile to phone buttons in a sequential manner. If the profile contains more directory and speed-dial numbers than there are buttons on the physical phone to which the profile is downloaded, the remaining numbers in the profile are ignored.

On Cisco Unified IP phones, line definitions are assigned to available extension buttons before speed-dial definitions, in sequential order, starting with the lowest directory number first.

After creating or modifying a profile, use the **reset** (voice logout-profile or voice user-profile) command to reset all phones associated with the profile being configured to propagate the changes.

Type **?** to list valid options for the **type** keyword. The following options are valid at the time that this document was written:

- **beep-ring**

Beep but no ring. Audible ring is suppressed for incoming calls but call-waiting beeps are allowed. Visible cues are the same as those for a normal ring.

- **feature-ring**

Differentiates incoming calls on a special line from incoming calls on other lines on the phone. The feature-ring cadence is a triple pulse, as opposed to a single-pulse ring for normal internal calls and a double-pulse ring for normal external calls.

- **monitor-ring**

A line button that is configured for monitor mode on one phone provides visual line status for a line that also appears on another phone. When monitor mode is set for a button with a shared line, the line status indicates that the shared line is either idle or in use. The line and line button are available in monitor mode for visual status only. Calls cannot be made or received using a line button that has been set in monitor mode. Incoming calls on a line button that is in monitor mode do not ring and do not display caller ID or call-waiting caller ID. Monitor mode is intended to be used only in the context of shared lines so that one user, such as a receptionist, can visually monitor the in-use status of several users' phone extensions (for example, as a busy-lamp field).

The line button for a monitored line can be used as a direct-station-select for a call transfer when the monitored line is in an idle state. In this case, the phone user who transfers a call from a normal line can press the Transfer button and then press the line button of the monitored line, causing the call to be transferred to the phone number of the monitored line.

- **silent-ring**

You can configure silent ring on any type of phone. However, you typically set silent ring only on buttons of a phone with multiple lines, such as a Cisco Unified IP Phone 7940 or Cisco Unified IP Phone 7960 and 7960G. The only visible cue is a flashing icon in the phone display.

If you configure a button to have a silent ring, you will not hear a call-waiting beep or call-waiting ring regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting beep or call-waiting ring.

Note In Cisco IOS Release 12.4(4)XC and later releases, the silent ringing behavior is overridden during active night-service periods. Silent ringing does not apply during designated night-service periods when the **s** keyword is used.

- **overlay**

Overlay lines are directory numbers that share a single line button on a multibutton phone. When more than one incoming call arrives on lines that are set on a single button, the line (ephone-dn) that is the left most in the **number** command list is the primary line and is given the highest priority. If this call is answered by another phone or if the caller hangs up, the phone selects the next line in its overlay set to present as the ringing call. The caller ID display updates to show the caller ID for the currently presented call.

Directory numbers that are part of an overlay set can be single-line directory numbers or dual-line directory numbers, but the set must contain either all single-line or all dual-line directory numbers, and not a mixture of the two.

The primary directory number on each phone in a shared-line overlay set should be a unique ephone-dn. The unique ephone-dn guarantees that the phone will have a line available for outgoing calls, and ensures that the phone user can obtain dial-tone even when there are no idle lines available in the rest of the shared-line overlay set. Use a unique directory number in this manner to provide for a unique calling party identity on outbound calls made by the phone so that the called user can see which specific phone is calling.

The name of the first directory number in the overlay set is not displayed because it is the default directory number for calls to the phone, and the name or number is permanently displayed next to the phone's button. For example, if there are ten numbers in an overlay set, only the last nine numbers are displayed when calls are made to them.

- **cw-overlay**

The same configuration is used for overlaid lines both with and without call waiting.

Directory numbers can accept call interruptions, such as call waiting, by default. For call waiting to work, the default must be active. To ensure that this is the case, remove the **no call-waiting beep accept** command from the configurations of directory numbers for which you want to use call waiting.

Directory numbers that are part of a cw-overlay set can be single-line directory numbers or dual-line directory numbers, but the set must contain either all single-line or all dual-line directory numbers, and not a mixture of the two.

The Cisco Unified IP Phone 7931G cannot support overlays that contain directory numbers that are configured for dual-line mode.

Examples

The following example shows the configuration for a voice-user profile to be downloaded when the a phone user logs into a Cisco Unified IP phone that is enabled for extension mobility. The lines and speed-dial buttons in this profile that are configured on an IP phone after the user logs in depend on phone type. For example, if the user logs into a Cisco Unified IP Phone 7970, all buttons are configured according to voice-user profile1. However, if the phone user logs into a Cisco Unified IP Phone 7960, all six lines are mapped to phone buttons, and the speed dial is ignored because there is no button available for speed dial.

```
pin 12345
user me password pass123
number 2001 type silent-ring
```

```
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones to which a particular logout-profile or user-profile is downloaded.

num-buttons

To set the number of line buttons supported by a phone type, use the **num-buttons** command in ephone-type configuration mode. To reset to the default, use the **no** form of this command.

num-buttons *number*
no num-buttons

Syntax Description

number	Number of line buttons. Range: 1 to 100. Default: 0. See the table for the number of buttons supported by each phone type.

Command Default

No line buttons are supported by the phone type.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command defines the number of line buttons supported by the type of phone being added with an ephone-type template.

Table 12: Supported Values for Ephone-Type Commands

Supported Device	device-id	device-type	num-buttons	max-presentation
Cisco Unified IP Phone 6901	547	6901	1	1
Cisco Unified IP Phone 6911	548	6911	1	10
Cisco Unified IP Phone 7915 Expansion Module with 12 buttons	227	7915	12	0 (default)
Cisco Unified IP Phone 7915 Expansion Module with 24 buttons	228	7915	24	0
Cisco Unified IP Phone 7916 Expansion Module with 12 buttons	229	7916	12	0
Cisco Unified IP Phone 7916 Expansion Module with 24 buttons	230	7916	24	0
Cisco Unified Wireless IP Phone 7925	484	7925	6	4
Cisco Unified IP Conference Station 7937G	431	7937	1	6
Nokia E61	376	E61	1	1

Examples

The following example shows that 1 line button is specified for the Nokia E61 when creating the ephone-type template.

```
Router(config)# ephone-type E61
Router(config-ephone-type)# num-buttons 1
```

Related Commands

Command	Description
device-id	Specifies the device ID for a phone type.
max-presentation	Sets the number of call presentation lines supported by a phone type.
type	Assigns the phone type to an SCCP phone.

num-line

To define the maximum number of lines supported by new phone, use the **num-line** command in voice register pool-type mode. To remove the lines configured, use the **no** form of this command.

num-line *max-line*
nonum-line*max-line*

Syntax Description

description	Specific the number of lines supported by the phone model. Range is 1-114.

Command Default

The default value of the addons is 1. When the **reference-pooltype** command is configured, the number of lines supported by the reference phone is inherited.

Command Modes

Voice Register Pool Configuration (config-register-pool)

Command History

Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to define the maximum number of lines for a Cisco Unified SIP IP phone on Cisco Unified CME. When you use the **no** form of this command, the inherited properties of the reference phone takes precedence over the default value.

Cisco Unified CME

The following example shows how to enter voice register pool-type configuration mode and define the maximum number of lines for a Cisco Unified SIP IP phone:

```
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# num-line 5
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.

Cisco Unified CME Commands: O

olsontimezone

To set the Olson Timezone so that the correct local time is displayed on Cisco Unified SCCP IP phones or Cisco Unified SIP IP phones, use the **olsontimezone** command in telephony-service or voice register global configuration mode, respectively. To return to the default, use the **no** form of this command.

olsontimezone *timezone* **version** *number*
no olsontimezone

Syntax Description

timezone	Olson Timezone names, which include the area (name of continent or ocean) and location (name of a specific location within that region, usually cities or small islands).
version *number*	Version of the tzupdater.jar or TzDataCSV.csv file. The version indicates whether the file needs to be updated or not. **Note** In Cisco Unified CME 9.0, the latest version is 2010o.

Command Default

No Olson Timezone is set.

Command Modes

Telephony-service configuration (config-telephony)

Voice register global configuration (config-register-global)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **olsontimezone** command in either telephony-service or voice register global configuration mode, with the current version of Oracle's Olson Timezone updater tool, tzupdater.jar, to set the correct Olson Timezone.

For Cisco Unified 3911 and 3951 SIP IP phones and Cisco Unified 6921, 6941, 6961, and 6945 SCCP and SIP IP phones, the correct Olson Timezone updater file is TzDataCSV.csv. The TzDataCSV.csv file is created based on the tzupdater.jar file.

To set the correct time zone, you must determine the Olson Timezone area/location where the Cisco Unified CME is located and download the latest tzupdater.jar or TzDataCSV.csv to a TFTP server (such as flash or slot 0) that is accessible to the Cisco Unified CME.

After a complete reboot, the phone checks if the version of its configuration file is earlier or later than 2010o. If it is earlier, the phone loads the latest tzupdater.jar and uses that updater file to calculate the Olson Timezone.

To make the Olson Timezone feature backward compatible, both the **time-zone** and **timezone** commands are retained as legacy time zones. Because the **olsontimezone** command covers approximately 500 time zones (Version 2010o of the tzupdater.jar file supports approximately 453 Olson Timezone IDs.), this command takes precedence when either the **time-zone** or the **timezone** command (that covers a total of 90 to100 time zones only) is present at the same time as the **olsontimezone** command.

Examples

The following example shows 7:29 p.m. as the time set on a Cisco Unified 7961 SCCP IP phone in Buenos Aires on May 13, 2011:

```
Router(config)# tftp-server flash:tzupdater.jar
Router(config)# tftp-server flash:TzDataCSV.csv
Router(config)# telephony-service
Router(config-telephony)# olsontimezone America/Argentina/Buenos Aires version 2010o
Router(config-telephony)# create cnf-files
Router(config-telephony)# time-zone 21
Router(config-telephony)# exit
Router(config)# clock timezone CST -6
Router(config)# clock summer-time date 12 October 2010 2:00 26 April 2011 2:00
Router(config)# exit
Router# clock set 19:29:00 13 May 2011
Router# configure terminal
Router(config)# telephony-service
Router(config-telephony)# reset
```

The following example shows 3:25 p.m. as the time set on a Cisco Unified 6921 SIP IP phone in Buenos Aires on November 17, 2011:

```
Router(config)# tftp-server slot0:tzupdater.jar
Router(config)# tftp-server slot0:TzDataCSV.csv
Router(config)# voice register global
Router(config-register-global)# olsontimezone America/Argentina/Buenos Aires version 2010o
Router(config-register-global)# create profile
Router(config-register-global)# timezone 21
Router(config-register-global)# exit
Router(config)# clock timezone CST -6
Router(config)# clock summer-time date 12 October 2010 2:00 26 April 2011 2:00
Router(config)# exit
Router# clock set 15:25:00 17 November 2011
Router# configure terminal
Router(config)# voice register global
Router(config-register-global)# reset
```

Related Commands

Command	Description
time-zone	Sets the time zone so that the correct local time is displayed on Cisco Unified SCCP IP phones in a Cisco Unified CME system.
timezone	Sets the time zone used for Cisco Unified SIP IP phones in a Cisco Unified CME system.

olsontimezone

To set the Olson Timezone so that the correct local time is displayed on Cisco Unified SCCP IP phones or Cisco Unified SIP IP phones, use the **olsontimezone** command in telephony-service or voice register global configuration mode, respectively. To return to the default, use the **no** form of this command.

olsontimezone *timezone* **version** *number*
no olsontimezone

Syntax Description

timezone	Olson Timezone names, which include the area (name of continent or ocean) and location (name of a specific location within that region, usually cities or small islands).
version *number*	Version of the tzupdater.jar or TzDataCSV.csv file. The version indicates whether the file needs to be updated or not. **Note** In Cisco Unified CME 9.0, the latest version is 2010o.

Command Default

No Olson Timezone is set.

Command Modes

Telephony-service configuration (config-telephony)

Voice register global configuration (config-register-global)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **olsontimezone** command in either telephony-service or voice register global configuration mode, with the current version of Oracle's Olson Timezone updater tool, tzupdater.jar, to set the correct Olson Timezone.

For Cisco Unified 3911 and 3951 SIP IP phones and Cisco Unified 6921, 6941, 6961, and 6945 SCCP and SIP IP phones, the correct Olson Timezone updater file is TzDataCSV.csv. The TzDataCSV.csv file is created based on the tzupdater.jar file.

To set the correct time zone, you must determine the Olson Timezone area/location where the Cisco Unified CME is located and download the latest tzupdater.jar or TzDataCSV.csv to a TFTP server (such as flash or slot 0) that is accessible to the Cisco Unified CME.

After a complete reboot, the phone checks if the version of its configuration file is earlier or later than 2010o. If it is earlier, the phone loads the latest tzupdater.jar and uses that updater file to calculate the Olson Timezone.

To make the Olson Timezone feature backward compatible, both the **time-zone** and **timezone** commands are retained as legacy time zones. Because the **olsontimezone** command covers approximately 500 time zones (Version 2010o of the tzupdater.jar file supports approximately 453 Olson Timezone IDs.), this command takes precedence when either the **time-zone** or the **timezone** command (that covers a total of 90 to100 time zones only) is present at the same time as the **olsontimezone** command.

Examples

The following example shows 7:29 p.m. as the time set on a Cisco Unified 7961 SCCP IP phone in Buenos Aires on May 13, 2011:

```
Router(config)# tftp-server flash:tzupdater.jar
Router(config)# tftp-server flash:TzDataCSV.csv
Router(config)# telephony-service
Router(config-telephony)# olsontimezone America/Argentina/Buenos Aires version 2010o
Router(config-telephony)# create cnf-files
Router(config-telephony)# time-zone 21
Router(config-telephony)# exit
Router(config)# clock timezone CST -6
Router(config)# clock summer-time date 12 October 2010 2:00 26 April 2011 2:00
Router(config)# exit
Router# clock set 19:29:00 13 May 2011
Router# configure terminal
Router(config)# telephony-service
Router(config-telephony)# reset
```

The following example shows 3:25 p.m. as the time set on a Cisco Unified 6921 SIP IP phone in Buenos Aires on November 17, 2011:

```
Router(config)# tftp-server slot0:tzupdater.jar
Router(config)# tftp-server slot0:TzDataCSV.csv
Router(config)# voice register global
Router(config-register-global)# olsontimezone America/Argentina/Buenos Aires version 2010o
Router(config-register-global)# create profile
Router(config-register-global)# timezone 21
Router(config-register-global)# exit
Router(config)# clock timezone CST -6
Router(config)# clock summer-time date 12 October 2010 2:00 26 April 2011 2:00
Router(config)# exit
Router# clock set 15:25:00 17 November 2011
Router# configure terminal
Router(config)# voice register global
Router(config-register-global)# reset
```

Related Commands

Command	Description
time-zone	Sets the time zone so that the correct local time is displayed on Cisco Unified SCCP IP phones in a Cisco Unified CME system.
timezone	Sets the time zone used for Cisco Unified SIP IP phones in a Cisco Unified CME system.

overlap-signal

To configure overlap dialing in SCCP or SIP IP phones, use the overlap-signal command in ephone, ephone-template, telephony-service, voice register pool, voice register global, or voice register template configuration mode.

overlap-signal

Syntax Description

This command has no arguments or keywords.

Command Default

Overlap-signal is disabled.

Command Modes

Call-manager-fallback
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Telephony-service configuration (config-telephony)
Voice register pool (config-register-pool)
Voice register global configuration (config-register-global)
Voice register template (config-register-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5 Cisco Unified SRST 8.5	This command was introduced.

Usage Guidelines

SCCP IP phones

In SCCP IP phones, overlap dialing is enabled when the overlap signal command is configured in ephone, ephone-template, and telephony-service configurations modes.

SIP IP phones

In SIP IP Phones, overlap dialing is enabled when the overlap signal command is configured in voice register pool, voice register global, and voice register template configuration modes.

Cisco Unified SRST

In Cisco Unified SRST, overlap dialing is enabled on SCCP IP phones when overlap signal command is configured in call-manager-fallback configuration mode.

Examples

The following example shows overlap-signal enabled on SCCP phones:

```
Router# show running config
!
!
telephony-service
 max-ephones 25
 max-dn 15
 load 7906 SCCP11.8-5-3S.loads
 load 7911 SCCP11.8-5-3S.loads
 load 7921 CP7921G-1.3.3.LOADS
 load 7941 SCCP41.8-5-3S.loads
 load 7942 SCCP42.8-5-3S.loads
 load 7961 SCCP41.8-5-3S.loads
```

```
 load 7962 SCCP42.8-5-3S.loads
 max-conferences 12 gain -6
 web admin system name cisco password cisco
 transfer-system full-consult
 create cnf-files version-stamp Jan 01 2002 00:00:00
 overlap-signal
!
ephone-template  1
 button-layout 1 line
 button-layout 3-6 blf-speed-dial
!
ephone-template  9
 feature-button 1 Endcall
 feature-button 3 Mobility
!
!
ephone-template  10
 feature-button 1 Park
 feature-button 2 MeetMe
 feature-button 3 CallBack
 button-layout 1 line
 button-layout 2-4 speed-dial
 button-layout 5-6 blf-speed-dial
 overlap-signal
!
ephone  10
 device-security-mode none
 mac-address 02EA.EAEA.0010
 overlap-signal
```

!

The following example shows overlap-signal configured in voice register global and voice register pool 10:

```
Router#show running config
!
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 media flow-around
 allow-connections sip to sip
!
voice class media 10
 media flow-around
!
!
voice register global
 max-pool 10
 overlap-signal
!
voice register pool  5
 overlap-signal
!
!
!
```

The following example shows overlap-signal configured in call-manager-fallback mode:

```
Router# show run | sec call-manager
call-manager-fallback
```

```
max-conferences 12 gain -6
transfer-system full-consult
overlap-signal
```

overwrite-dyn-stats (voice hunt-group)

To overwrite statistics of previously joined dynamic agent with statistics of newly joined dynamic agents for voice hunt group, use the **overwrite-dyn-stats** command in voice hunt-group configuration mode. To remove the configuration, use the **no** form of this command.

overwrite-dyn-stats
no overwrite-dyn-stats

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this command is disabled.

Command Modes

voice hunt group configuration (config-voice-hunt-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

This command is configured to overwrite statistics of previously joined dynamic agent with statistics of newly joined dynamic agents for voice hunt group. To remove the configuration, use the no form of this command. The statistics for the first 32 members (both dynamic and static members) joining in an hour are collected in the 32 statistic slots allotted. If any of the static members logout and login during the hour, that member is allotted the same slot as previous. In scenarios where free slots are available, free slots are used to write statistics of the newly joined dynamic agent. Once all the 32 slots are exhausted and a new dynamic member tries to join within the same hour, the **overwrite-dyn-stats** CLI takes effect. Using the CLI, the hunt statistic slot for the first dynamic member that joined the hunt-group is overwritten with the statistics of the newly joined dynamic member. The overwriting for statistics will continue at the same slot.

Examples

The following example shows how the voice hunt group overwrite-dyn-stats option is enabled:

```
Router(config)# voice hunt-group 1 parallel
Router(config-voice-hunt-group)# overwrite-dyn-stats
```

Cisco Unified CME Commands: P

paging

To define an extension (ephone-dn) as a paging extension that can be called to broadcast an audio page to a set of Cisco IP phones, use the **paging** command in ephone-dn configuration mode. To disable this feature, use the **no** form of this command.

paging [**ip** *multicast-address* **port** *udp-port-number*]
no paging [**ip**]

Syntax Description

ip *multicast-address*	(Optional) Uses an IP multicast address to multicast voice packets for audio paging; for example, 239.0.1.1. Note that IP phones do not support multicast at 224.x.x.x addresses. Default is that multicast is not used and IP phones are paged individually using IP unicast transmission (up to ten phones).
port *udp-port-number*	(Optional) Uses this UDP port for the multicast. Range is from 2000 to 65535. Default is 2000.

Command Default

No paging number is established.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

To configure a set of phones to receive an audio page, follow these steps:

1. Use the **paging** command in ephone-dn configuration mode to define a number that people can dial to send a page. The following example defines a paging-dn tag (21) and extension number (34455) to dial to send a page.

```
ephone-dn 21
 paging
 number 34455
```

1. Use the **paging-dn** command in ephone configuration mode to assign the same paging-dn tag that you defined in Step 1 to all the phones that you want to receive the page. This set of phones is called a "paging set." You can have more than one paging set in a Cisco CME system. The following example assigns the paging-dn tag from Step 1 (21) to two phones (3 and 4) so that they will receive audio pages.

```
ephone 3
 paging-dn 21
ephone 4
 paging-dn 21
```

The **paging** command configures an ephone-dn as an extension that people can dial to broadcast audio pages to a specified set of idle Cisco IP phones. The extension associated with this command does not appear on any ephone; it is a "dummy" extension. The dn-tag associated with this extension becomes the paging-dn tag for this paging set.

When a person dials the number assigned to the dummy extension and speaks into the phone, the audio stream is sent as a page to the paging set (the set of all phones that have been configured with this paging-dn tag as an argument to the **paging-dn** command). Idle phones in the paging set automatically answer the paging call in one-way speakerphone mode. Paging sets can be joined into a single combined paging group with the **paging group** command.

The optional **ip** keyword and *multicast-address* argument define a paging multicast address for this paging set. If an IP multicast address is not configured, IP phones are paged individually using IP unicast transmission (to a maximum of ten IP phones). The recommended operation is with an IP multicast address. When multiple paging-dn tags are configured using the **paging** command, each paging-dn tag should use a unique IP multicast address.

Note IP phones do not support multicast at 224.x.x.x addresses.

Each ephone-dn and paging-dn tag that is used for paging can support a maximum of ten distinct targets (IP addresses and interfaces). A multicast address counts as a single target for each physical interface in use (regardless of the number of phones connected via the interface). Paging using a single IP multicast address that requires output on three different Ethernet interfaces represents use of three counts out of the maximum ten. Each unicast target counts as a single target, such that paging that does not use multicast at all is limited to paging ten phones. For example, ten IP phones paged through multicast on Fast Ethernet interface 0/1.1 plus five IP phones paged through multicast on Fast Ethernet interface 0/1.2 are counted as two targets.

For simultaneous paging to more than one paging ephone-dn, Cisco recommends that you use different IP multicast addresses (not just different port numbers) for paging configuration.

Examples

The following example creates a paging extension number that uses IP multicast paging:

```
Router(config)# ephone-dn 20
Router(config-ephone-dn) number 2000
Router(config-ephone-dn) paging ip 239.0.1.1 port 2000
```

A more complete configuration example follows, in which paging sets 20 and 21 are created. Pages to extension 2000 are multicast to ephones 1 and 2. Pages to extension 2001 are multicast to ephones 3 and 4.

```
ephone-dn 1
 number 2345
ephone-dn 2
 number 2346
ephone-dn 3
 number 2347
ephone-dn 4
 number 2348
ephone-dn 20
 number 2000
 paging ip 239.0.1.20 port 2000
ephone-dn 21
 number 2001
 paging ip 239.0.1.21 port 2000
ephone 1
 button 1:1
 paging-dn 20
ephone 2
 button 1:2
```

```
 paging-dn 20
ephone 3
 button 1:3
 paging-dn 21
ephone 4
 button 1:4
 paging-dn 21
```

Related Commands

Command	Description
paging-dn	Assigns audio paging reception capability to a Cisco IP phone.
paging group	Combines two or more paging sets into a combined paging group.

paging group

To create a combined paging group from two or more previously established paging sets, use the **paging group** command in ephone-dn configuration mode. To remove a paging group, use the **no** form of this command.

paging group*paging-dn-tag, paging-dn-tag...*

no paging group

Syntax Description

paging-dn-tag	Comma-separated list of paging-dn-tags (unique sequence numbers associated with paging ephone-dns) that have previously been associated with the paging extension of a paging set using the **paging-dn** or **paging-dn (voice register)** command. You can include up to ten paging-dn-tags separated by commas. For example, 4, 6, 7, 8.

Command Default

Paging is disabled on all Cisco IP phones.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
15.2(2)T	Cisco Unifide CME 9.0	This command was modified to include voice register pools in the ephone-dn and paging groups.

Usage Guidelines

Use this command to combine previously defined sets of phones associated with individual paging extensions (ephone-dns) into a combined group to enable a single page to be sent to large numbers of phones at once. To remove a paging group, use the **no** form of the command. All paging-dn tags included in the list must have already been defined as paging-dns using the **paging** or **paging-dn (voice register)** command.

The use of paging groups not only allows phones to participate in a small local paging set (for example, paging to four phones in a company’s shipping and receiving department) but also supports company-wide paging when needed (for example, by combining the paging sets for shipping and receiving with the paging sets for accounting, customer support, and sales into a single paging group).

Note The correct paging port for the paging-dn of Cisco Unified SIP IP phones in the **paging** command is an even number from 20480 to 32768. If you enter a wrong port number, a SIP REFER message request is sent to the IP phone but the Cisco Unified SIP IP phone is not paged.

Examples

In the following example, paging sets 20 and 21 are defined and then combined into paging group 22. Paging set 20 has a paging extension of 2000. When someone dials extension 2000 to deliver a page, the page is sent to Cisco IP phones (ephones) 1 and 2. Paging set 21 has a paging extension of 2001. When someone dials extension 2001 to deliver a page, the page is sent to ephones 3 and 4.

Paging group 22 combines sets 20 and 21, and when someone dials its paging extension, 2002, the page is sent to all the phones in both sets and to ephone 5, which is directly subscribed to the combined paging group.

```
ephone-dn 20
 number 2000
 paging ip 239.0.1.20 port 2000
ephone-dn 21
 number 2001
 paging ip 239.0.1.21 port 2000
ephone-dn 22
 number 2002
 paging ip 239.0.2.22 port 2000
 paging group 20,21
ephone 1
 button 1:1
 paging-dn 20
ephone 2
 button 1:2
 paging-dn 20
ephone 3
 button 1:3
 paging-dn 21
ephone 4
 button 1:4
 paging-dn 21
ephone 5
 button 1:5
 paging-dn 22
```

The following example shows how the **paging group** command is used to configure combined paging groups composed of ephone and voice register directory numbers.

The first set of configuration tasks shows how to configure a combined paging group composed of Cisco Unified SCCP IP phone directory numbers only.

When extension 2000 is dialed, a page is sent to ephones 1 and 2 (first single paging group). When extension 2001 is dialed, a page is sent to ephones 3 and 4 (second single paging group). Finally, when extension 2002 is dialed, a page is sent to ephones 1, 2, 3, 4, and 5, producing the combined paging group (composed of the first single paging group, the second single paging group, and ephone 5).

Ephones 1 and 2 are included in paging ephone-dn 22 through the membership of ephone-dn 20 as paging group 20 in the combined paging group. Ephones 3 and 4 are included in paging ephone-dn 22 through membership of ephone-dn 21 as paging group 21 in the combined paging group. Ephone 5 is directly subscribed to paging-dn 22.

```
ephone-dn 20
 number 2000
 paging ip 239.0.1.20 port 20480
ephone-dn 21
 number 2001
 paging ip 239.1.1.21 port 20480
ephone-dn 22
 number 2002
 paging ip 239.1.1.22 port 20480
 paging group 20,21
ephone-dn 6
 number 1103
```

```
ephone-dn 7
 number 1104
ephone-dn 8
 number 1105
ephone-dn 9
 number 1199
ephone-dn 10
 number 1198
ephone 1
 mac-address 1234.8903.2941
 button 1:6
 paging-dn 20
ephone 2
 mac-address CFBA.321B.96FA
 button 1:7
 paging-dn 20
ephone 3
 mac-address CFBB.3232.9611
 button 1:8
 paging-dn 21
ephone 4
 mac-address 3928.3012.EE89
 button 1:9
 paging-dn 21
ephone 5
 mac-address BB93.9345.0031
 button 1:10
 paging-dn 22
```

The second set of configuration tasks shows how Cisco Unified SIP IP phone directory numbers can be configured and added to the previously established paging groups of the first set of configuration tasks to form a new combined paging group composed of ephone and voice register directory numbers.

When extension 2000 is dialed, a page is sent to ephones 1 and 2 and voice register pools 1 and 2 (new first single paging group). When extension 2001 is dialed, a page is sent to ephones 3 and 4 and voice register pools 3 and 4 (newsecond single paging group). Finally, when extension 2002 is dialed, a page is sent to ephones 1, 2, 3, 4, and 5 and voice register pools 1, 2, 3, 4, and 5 (new combined paging group).

Ephones 1 and 2 and voice register pools 1 and 2 are included in paging ephone-dn 22 through the membership of ephone-dn 20 as paging group 20 in the combined paging group. Ephones 3 and 4 and voice register pools 3 and 4 are included in paging ephone-dn 22 through membership of ephone-dn 21 as paging group 21 in the combined paging group. Ephone 5 and voice register pool 5 are directly subscribed to paging-dn 22.

```
voice register dn 1
 number 1201
voice register dn 2
 number 1202
voice register dn 3
 number 1203
voice register dn 4
 number 1204
voice register dn 5
 number 1205
voice register pool 1
 id mac 0019.305D.82B8
 type 7961
 number 1 dn 1
 paging-dn 20
voice register pool  2
```

```
 id mac 0019.305D.2153
 type 7961
 number 1 dn 2
 paging-dn 20
voice register pool  3
 id mac 1C17.D336.58DB
 type 7961
 number 1 dn 3
 paging-dn 21
voice register pool  4
 id mac 0017.9437.8A60
 type 7961
 number 1 dn 4
 paging-dn 21
voice register pool  5
 id mac 0016.460D.E469
 type 7961
 number 1 dn 5
 paging-dn 22
```

Related Commands

Command	Description
paging	Creates a paging extension (ephone-dn) that can be called in order to broadcast an audio page to a group of Cisco IP phones.
paging-dn	Assigns a paging extension (paging-dn) to a Cisco IP phone.
paging-dn (voice register)	Registers a Cisco Unified SIP IP phone to an ephone-dn paging directory number.

paging-dn

To create a paging extension (paging-dn) to receive audio pages on a Cisco Unified IP phone in a Cisco Unified CME system, use the **paging-dn** command in ephone or ephone-template configuration mode. To disable this feature, use the **no** form of this command.

paging-dn *paging-dn-tag* {**multicast** | **unicast**}
paging-dnno

Syntax Description

paging-dn-tag	Dn-tag of an ephone-dn that was designated as a paging ephone-dn with the **paging** command.
multicast	Uses multicast if available. By default, audio paging is transmitted to the Cisco Unified IP phone using multicast.
unicast	Forces unicast paging for this phone. This keyword indicates that the Cisco Unified IP phone is not capable of receiving audio paging through multicast and requests that all pages to this phone be sent through unicast. The maximum number of phones supported through unicast is ten.

Command Default

Paging is disabled on all Cisco Unified IP phones.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

To configure a set of phones to receive an audio page, follow these steps:

1. Use the **paging** command in ephone-dn configuration mode to define a number that people can dial to send a page. The following example defines a paging-dn tag (21) and extension number (34455) to dial to send a page.

```
ephone-dn 21
 paging
 number 34455
```

1. Use the **paging-dn** command in ephone configuration mode to assign the same paging-dn tag that you defined in Step 1 to all the phones that you want to receive the page. This set of phones is called a “paging set.” You can have more than one paging set in a Cisco Unified CME system. The following example assigns the paging-dn tag from Step 1 (21) to two phones (3 and 4) so that they will receive audio pages.

```
ephone 3
 paging-dn 21
ephone 4
 paging-dn 21
```

This command creates a paging extension (paging-dn) associated with an IP phone. Each phone can support only one paging-dn extension. This extension does not occupy a phone button and is therefore not configured on the phone with the **button** command. The paging-dn allows the phone to automatically answer audio pages in one-way speakerphone mode. There is no press-to-answer option as there is with an intercom extension.

The *paging-dn-tag* argument in this command takes the value of the dn-tag of an extension (ephone-dn) that has been made a paging ephone-dn using the **paging** command. This command is the extension that callers dial to deliver an audio page. All of the phones that are going to receive the same audio pages are configured with the same *paging-dn-tag*. These phones form a paging set.

An IP phone can belong to only one paging set, but any number of phones can belong to the same paging set using multicast. There can be any number of paging sets in a Cisco Unified CME system, and paging sets can be joined to create a combined paging group using the **paging group command.** For example, you may create separate paging sets for each department (sales, support, shipping) and combine them into a single combined paging group (all departments). Only single-level grouping is supported (no support for groups of groups).

Normal phone calls that are received while an audio page is in progress interrupt the page.

The paging mechanism supports audio distribution using IP multicast, replicated unicast, and a mixture of both (so that multicast is used where possible, and unicast is used with specific phones that cannot be reached through multicast).

Note For unicast paging to all phones, omit the IP multicast address in the ephone-dn configuration. For unicast paging to a specific phone using an ephone-dn configured for multicast, add the **unicast** keyword as part of the **paging-dn** command in ephone configuration mode.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example creates paging number 5001 on ephone-dn 22 and adds ephone 4 as a member of the paging set. Multicast is set for the paging-dn. Note that IP phones do not support multicast at 224.x.x.x addresses.

```
ephone-dn 1
 number 5123
ephone-dn 22
 name Paging Shipping
 number 5001
 paging ip 239.1.1.10 port 2000
ephone 4
 mac-address 0030.94c3.8724
 button 1:1
 paging-dn 22 multicast
```

Related Commands

	Description
ephone-template (ephone)	Applies a template to an ephone configuration.

	Description
number	Configures a valid number for the Cisco Unified IP phone.
paging	Creates a paging extension (ephone-dn) that can be called in order to broadcast an audio page to a group of Cisco Unified IP phones.
paging group	Combines two or more paging sets into a combined paging group.

paging-dn (voice register)

To register a Cisco Unified SIP IP phone to an ephone-dn paging directory number (DN), use the **paging-dn** command in voice register pool or voice register template configuration mode. To unregister the Cisco Unified SIP IP phone from the paging directory number, use the **no** form of this command.

paging-dn *paging-dn-tag* {**multicast** | **unicast**}
no paging-dn

Syntax Description

paging-dn-tag	Ephone-dn tag designated as the paging ephone-dn to which a Cisco Unified SIP IP phone is registered.
multicast	Transmits audio paging to the Cisco Unified IP phone using multicast. This is the default.
unicast	Transmits audio paging to the Cisco Unified IP phone using unicast.This indicates that the Cisco Unified IP phone is not capable of receiving audio paging through multicast and requests that all pages to this phone be sent through unicast. The maximum number of phones supported through unicast is 12.

Command Default

The Cisco Unified SIP IP phone is not registered to an ephone-dn paging DN and paging is disabled.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration (config-register-temp)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

The **paging-dn** command applies to both voice register pool and voice register template configuration modes. When voice register pool is configured with the template and paging is configured in voice register pool configuration mode, paging in voice register pool configuration mode has higher precedence over paging in voice register template configuration mode.

The correct paging port for the paging-dn of Cisco Unified SIP IP phones in the **paging** command is an even number from 20480 to 32768. If you enter a wrong port number, a SIP REFER message request is sent to the IP phone but the Cisco Unified SIP IP phone is not paged.

Examples

The following example shows how the Cisco Unified 7961 SIP IP phone is registered to both paging-dns 251 and 252:

```
ephone-dn  2  dual-line
 number 60012
ephone-dn  250
number 7770
 paging ip 239.1.1.0 port 20480
 paging group 251,252
ephone-dn  251
 number 7771
 paging ip 239.1.1.1 port 20480
ephone-dn  252
```

```
 number 7772
 paging ip 239.1.1.2 port 20480

ephone-dn  253
 number 7773
 paging ip 239.1.1.3 port 20480
ephone  2
 mac-address 001E.4A91.F27D
 paging-dn 252
 type 7961
 button  1:2
voice register dn  1
 number 60001
voice register dn  2
 number 60002
voice register pool  1
 id mac 0019.305D.82B8
 type 7961
 number 1 dn 1
 codec g711ulaw
 paging-dn 251
voice register pool  2
 id mac 0019.305D.2153
 type 7961
 number 1 dn 2
 codec g711ulaw
 paging-dn 252
```

Related Commands

Command	Description
paging-dn	Creates a paging extension to receive audio pages on a Cisco Unified SCCP IP phone in a Cisco Unified CME system.
paging group	Creates a combined paging group from two or more previously established paging sets.

param

To load and configure parameters in a package or a service (application) on the gateway, use the **param** command in application configuration mode. To reset a parameter to its default value, use the **no** form of this command.

param *param-name* [{**param max-retries** | **param passwd** | **param passwd-prompt filename** | **param user-prompt filename** | **param term-digit** | **param abort-digit** | **param max-digits**}]
no param *param-name*

Syntax Description

param-name	Name of the parameter.
param max-retries	(Optional) Number of attempts to re-enter account or password. Value ranges from 0-10, default value is 0.
param passwd	(Optional) Character string that defines a predefined password for authorization.
param passwd-prompt filename	(Optional) Announcement URL to request password input. filename defines the name and location of the audio filename to be used for playing the password prompt.
param user-prompt filename	(Optional) Announcement URL to request authorization code username. filename defines the name and location of the audio filename to be used for playing the username prompt.
param term-digit	Digit for terminating username or password digit input.
param abort-digit	Digit for aborting username or password digit input. Default value is *.
param max-digits	Maximum number of digits in a username or password. Range of valid value: 1 - 32. Default value is 32.

Command Default

No default behavior or value.

Command Modes

Application configuration

Command History

Release	Modification
12.3(14)T	This command was introduced.
15.1(3)T	This command was modified. The following keywords and arguments were added: param max-retries, param passwd, param passwd-prompt filename, param user-prompt filename, param term-digit, param max-digit.

Usage Guidelines

Use this command in application parameter configuration mode to configure parameters in a package or service. A package is a linkable set of C or Tcl functions that provide functionality invoked by applications or other packages. A service is a standalone application.

The parameters available for configuration differ depending on the package or service that is loaded on the gateway. The **param register** Tcl command in a service or package registers a parameter and provides a description and default values which allow the parameter to be configured using the CLI. The **param register** command is executed when the service or package is loaded or defined, along with commands such as **package provide**, which register the capability of the configured module and its associated scripts. You must configure and load the Tcl scripts for your service or package and load the package in order to configure its parameters. See the *Tcl IVR API Version 2.0 Programming Guide* for more information.

When a package or service is defined on the gateway, the parameters in that package or service become available for configuration when you use this command. Additional arguments and keywords are available for different parameters. To see a list of available parameters, enter **param** ?.

To avoid problems with applications or packages using the same parameter names, the *parameter namespace,* or *parameterspace* concept is introduced. When a service or a package is defined on the gateway, its parameter namespace is automatically defined. This is known as the service or package's local parameterspace, or "myparameterspace." When you use this command to configure a service or package's parameters, the parameters available for configuration are those contained in the local parameterspace. If you want to use parameter definitions found in different parameterspace, you can use the **paramspace** *parameter-namespace* command to map the package's parameters to a different parameterspace. This allows that package to use the parameter definitions found in the new parameterspace, in addition to its local parameterspace.

Use this command in Cisco Unified Communication Manager Express 8.5 and later versions to define the username and password parameters to authenticate packages for Forced Authorization Code (FAC)

When a predefined password is entered using the param passwd keyword, callers are not requested to enter a password. You must define a filename for user-prompt to play an audio prompt requesting the caller to enter a valid username (in digits) for authorization. Similarly, you must define a filename for passwd-prompt to play an audio prompt requesting the caller to enter a valid password (in digits) for authorization.

Examples

The following example shows how to configure a parameter in the httpios package:

```
application
package httpios
param paramA value4
```

Related Commands

Command	Description
call application voice	Defines the name of a voice application and specify the location of the Tcl or VoiceXML document to load for this application.
param account-id-method	Configures an application to use a particular method to assign the account identifier.
param convert-discpi-after-connect	Enables or disables conversion of a DISCONNECT message with Progress Indicator set to PROG_INBAND (PI=8) to a regular DISCONNECT message when the call is in the active state.
param event-log	Enables or disables logging for linkable Tcl functions (packages).
param language	Configures the language parameter in a service or package on the gateway.
param mode	Configures the call transfer mode for a package.

Command	Description
param pin-len	Defines the number of characters in the personal identification number (PIN) for an application.
param redirect-number	Defines the telephone number to which a call is redirected—for example, the operator telephone number of the service provider—for an application.
param reroutemode	Configures the call transfer reroutemode (call forwarding) for a package.
param retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
param security	Configures security for linkable Tcl functions (packages).
paramspace	Enables an application to use parameters from the local parameter space of another application.
param uid-length	Defines the number of characters in the UID for a package.
param warning-time	Defines the number of seconds of warning that a user receives before the allowed calling time expires.

param aa-hunt

To declare a Cisco Unified CME B-ACD menu number and associate it with the pilot number of an ephone hunt group, use the **param aa-hunt** command in application-parameter configuration mode. To remove the menu number and the ephone hunt group pilot number, use the **no** form of this command.

param aa-hunt *menu-number pilot-number*
no param aa-hunt *menu-number pilot-number*

Syntax Description

menu-number	Number that callers must dial to reach the pilot number of an ephone hunt group. The range is from 1 to 10. The default is 1.
pilot-number	Ephone hunt group pilot number.

Command Default

Menu number 1 is configured, but it is not associated with a pilot number**.**

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco CME 3.3	This command was introduced to replace the **call application voice aa-hunt** command.

Usage Guidelines

This command is used with Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. It is configured under the **service** command for the call-queue script.

Up to ten aa-hunt menu options, or hunt groups, are allowed per call-queue service. You can use any of the allowable numbers in any order.

This command associates a menu option with the pilot number of an ephone hunt group. When a caller presses the digit of a menu option that has been associated with an ephone hunt group using this command, the call is routed to the pilot number of the hunt group.

Menu options for B-ACD services can be set up in many ways. For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

The highest aa-hunt number that you establish using this command also automatically maps to zero (0) and can therefore be used to represent operator services to your callers. In the following example, callers can dial either 8 or 0 to reach aa-hunt8, the hunt group with the pilot number 8888.

```
application
 service queue flash:
  param aa-hunt1 1111
  param aa-hunt3 3333
  param aa-hunt8 8888
```

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

Examples

The following example configures a call-queue service called queue to associate three menu numbers with three pilot numbers of three ephone hunt groups:

- Pilot number 1111 for ephone hunt group 1 (sales)

- Pilot number 2222 for ephone hunt group 2 (customer service)
- Pilot number 3333 for ephone hunt group 3 (operator)

If a caller presses 2 for customer service, the call is transferred to 2222 and then is sent to the next available ephone-dn from the group of ephone-dns assigned to ephone hunt group 1: 2001, 2002, 2003, 2004, 2005, and 2006. The sequencing of ephone-dns within a hunt group is handled by the ephone hunt group itself, not by the B-ACD service. (Note that the configuration for ephone hunt groups used with Cisco Unified CME B-ACD services do not use the **final** command.)

```
ephone-hunt 1 peer
 pilot 1111
 list 1001, 1002, 1003, 1004, 1005, 1006, 1007, 1008, 1009, 1010
ephone-hunt 2 peer
 pilot 2222
 list 2001, 2002, 2003, 2004, 2005, 2006
ephone-hunt 3 peer
 pilot 3333
 list 3001, 3002, 3003, 3004
application
 service queue flash:
  param aa-hunt1 1111
  param aa-hunt2 2222
  param aa-hunt3 3333
.
.
.
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param aa-pilot

To assign a pilot number to a Cisco Unified CME B-ACD automated attendant (AA) service, use the **param aa-pilot** command in application-parameter configuration mode. To remove the AA pilot number, use the **no** form of this command.

param aa-pilot *aa-pilot-number*
no param aa-pilot *aa-pilot-number*

Syntax Description

aa-pilot-number	Telephone number that callers dial in order to reach this AA service.

Command Default

Cisco Unified CME B-ACD menu number 1 is configured, but it has no pilot number**.**

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice aa-pilot** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. Each AA has one AA pilot number, although there may be more than one AA used with a B-ACD service.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with two AAs, both in drop-through mode. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. Incoming POTS dial peers are established for both AA pilot numbers.

```
dial-peer voice 1010 pots
 service acdaa
 port 1/1/0
 incoming called-number 8005550121
dial-peer voice 1020 pots
 service aa-bcd
 port 1/1/1
 incoming called-number 8005550123
.
.
.
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
  param number-of-hunt-grps 2
  param queue-len 10
```

```
!
 service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550121
  param service-name callq
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 20
  param number-of-hunt-grps 1
  param drop-through-prompt _bacd_welcome.au
  param drop-through-option 2
  param second-greeting-time 45
  param handoff-string acdaa
  param max-time-call-retry 360
!
 service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550123
  param service-name callq
  param second-greeting-time 60
  param max-time-call-retry 180
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 5
  param handoff-string aa-bcd
  param drop-through-option 1
  param number-of-hunt-grps 1
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param call-retry-timer

To specify the time interval before each attempt to retry to connect a call to an ephone hunt group used with a Cisco CME B-ACD service, use the **param call-retry-timer** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param call-retry-timer *seconds*
no param call-retry-timer *seconds*

Syntax Description

seconds	Time that a call must wait before attempting or reattempting to transfer a call to an ephone hunt group pilot number, in seconds. Range is from 5 to 30 seconds.

Command Default

Default is 15 seconds.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice call-retry-timer command.**

Usage Guidelines

This command is used with Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service. A Cisco Unified CME B-ACD service can have more than one AA, and each AA can specify a different interval for retries to connect to ephone hunt group phones.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with two AAs. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. The first AA has a call-retry timer set to 10 seconds, and the second AA has a call-retry timer set to 5 seconds.

```
dial-peer voice 1010 pots
 service acdaa
 port 1/1/0
 incoming called-number 8005550121
dial-peer voice 1020 pots
 service aa-bcd
 port 1/1/1
 incoming called-number 8005550123
.
.
.
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
```

```
  param number-of-hunt-grps 2
  param queue-len 10
!
 service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550121
  param service-name callq
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 10
  param number-of-hunt-grps 1
  param drop-through-prompt _bacd_welcome.au
  param drop-through-option 2
  param second-greeting-time 45
  param handoff-string acdaa
  param max-time-call-retry 60
!
 service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550123
  param service-name callq
  param second-greeting-time 60
  param max-time-call-retry 180
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 5
  param handoff-string aa-bcd
  param drop-through-option 1
  param number-of-hunt-grps 1
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param co-did-max

To set the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) for use with the Direct Inward Dial (DID) Digit Translation Service, use the **param co-did-max** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param co-did-max *max-co-value*
no param co-did-max *max-co-value*

Syntax Description

max-co-value	Maximum value of digits coming from the CO. The digit string can be any length, but the string length must be the same in the **param co-did-min**, **param co-did-max**, **param store-did-min**, and **param store-did-max** commands.

Command Default

No maximum value is defined for the range of digits coming from the CO.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice co-did-max command.**
12.4(9)T	Cisco Unified CME 4.0	This command replaced the **call application voice co-did-max** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command defines the upper limit of the range of digits accepted from the CO when it is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service. This service provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

Examples

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the valid range of digits coming from the CO.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
  param co-did-max 79
```

```
param store-did-min 00
param store-did-max 79
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.
param co-did-min	Sets the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.
param store-did-max	Sets the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.
param store-did-min	Sets the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.

param co-did-min

To set the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the Direct Inward Dial (DID) Digit Translation Service, use the **param co-did-min** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param co-did-min *min-co-value*
no param co-did-min *min-co-value*

Syntax Description

min-co-value	Minimum value of digits coming from the CO. The digit string can be any length, but the string length must be the same in the **param co-did-max**, **param co-did-max**, **param store-did-min**, and **param store-did-max** commands.

Command Default

No minimum value is defined for the range of digits coming from the CO.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice co-did-min command.**
12.4(9)T	Cisco Unified CME 4.0	This command replaced the **call application voice co-did-min** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command defines the upper limit of the range of digits accepted from the CO when it is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service. This service provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

Examples

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the valid range of digits coming from the CO.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
  param co-did-max 79
```

```
param store-did-min 00
param store-did-max 79
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.
param co-did-max	Sets the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.
param store-did-max	Sets the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.
param store-did-min	Sets the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.

param dial-by-extension-option

To assign a menu number to an Cisco CME B-ACD option by which callers can directly dial known extension numbers, use the **param dial-by-extension-option** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param dial-by-extension-option *menu-number*
no param dial-by-extension-option *menu-number*

Syntax Description

menu-number	Menu option number to be associated with the dial-by-extension option. Range is from 1 to 9. There is no default.

Command Default

Dial-by-extension option is not assigned.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice dial-by-extension-option** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

This command allows you to designate a menu option number for callers to press if they want to dial an extension number that they already know. This command also enables the playing of the en_bacd_enter_dest.au audio file after a caller presses the dial-by-extension menu number. The default announcement in this audio file is “Please enter the extension number you want to reach.”

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with an AA called acd1, which has an AA pilot number of (800) 555-0121. The call-queue service used with this AA is named callq. Callers to (800) 555-0121 can press 1 to dial an extension number (**param dial-by-extension-option 1** under **service acd1**) or press 2 to be connected to the hunt group with the pilot number 5072 (**param aa-hunt2 5072** under **service callq**).

```
dial-peer voice 1010 pots
 service acd1
 port 1/1/0
 incoming called-number 8005550121
.
.
.
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt2 5072
```

```
  param number-of-hunt-grps 1
  param queue-len 10
!
 service acd1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param handoff-string acd1
  param service-name callq
  param aa-pilot 8005550121
  param number-of-hunt-grps 1
  param dial-by-extension-option 1
  param second-greeting-time 45
  param call-retry-timer 20
  param max-time-call-retry 360
  param max-time-vm-retry 2
  param voice-mail 5007
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param did-prefix

To set a prefix to add to digits coming from the PSTN Central Office (CO) to create valid extension numbers when using the Direct Inward Dial (DID) Digit Translation Service, use the **param did-prefix** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param did-prefix *did-prefix*
no param did-prefix *did-prefix*

Syntax Description

did-prefix	Prefix to add. Range is from 0 to 99.

Command Default

No prefix is defined.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME4.0	This command was introduced to replace the **call application voice did-prefix** command**.**
12.4(9)T	Cisco Unified CME4.0	This command replaced the **call application voice did-prefix** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service, which provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

Examples

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It specifies that a prefix of 5 should be applied to the digits coming from the CO in order to construct a valid extension number.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
  param co-did-max 79
  param store-did-min 00
  param store-did-max 79
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param drop-through-option

To assign the drop-through option to a Cisco Unified CME B-ACD auto-attendant (AA) application, use the **param drop-through option** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param drop-through-option *menu-number*
no param drop-through-option *menu-number*

Syntax Description

menu-number	Menu option number (aa-hunt number) to be associated with the drop-through option.

Command Default

Drop-through option is not assigned.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice drop-through-option** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

When an AA is configured for drop-through mode, the AA sends incoming calls directly to the call queue associated with the menu number specified in this command. Once in the queue, a caller hears ringback if an agent is available or music on hold (MOH) if all agents are busy. If a greeting prompt for drop-through mode is configured using the **param drop-through-prompt** command, a caller hears the prompt before being sent to the queue as described.

The menu option number is an aa-hunt number that is associated with an ephone hunt group using the **param aa-hunt** command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with two AAs, both in drop-through mode. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. Callers to (800) 555-0121 drop directly through to the hunt group with the pilot number 5072 after hearing the greeting prompt in the audio file named en_dto_welcome.au. Callers to (800) 555-0123 drop directly through to the hunt group with the pilot number 5071 without hearing any greeting.

```
dial-peer voice 1010 pots
 service acdaa
 port 1/1/0
 incoming called-number 8005550121
dial-peer voice 1020 pots
 service aa-bcd
 port 1/1/1
```

```
 incoming called-number 8005550123
.
.
.
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
  param number-of-hunt-grps 2
  param queue-len 10
!
 service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550121
  param service-name callq
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 20
  param number-of-hunt-grps 1
  param drop-through-prompt _bacd_dto_welcome.au
  param drop-through-option 2
  param second-greeting-time 45
  param handoff-string acdaa
  param max-time-call-retry 360
!
 service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550123
  param service-name callq
  param second-greeting-time 60
  param max-time-call-retry 180
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 5
  param handoff-string aa-bcd
  param drop-through-option 1
  param number-of-hunt-grps 1
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param drop-through-prompt

To associate an audio prompt file with the drop-through option for a Cisco Unified CME B-ACD automated attendant (AA) application, use the **param drop-through-prompt** command in application-parameter configuration mode. To disable the prompt, use the **no** form of this command.

param drop-through-prompt *audio-filename*
no param drop-through-prompt *audio-filename*

Syntax Description

audio-filename	Identifying part of the filename of the prompt to be played when calls for the drop-through option are answered.

Command Default

No prompt is designated for the drop-through option.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice drop-through-prompt** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

When an AA is configured for drop-through mode, the AA sends incoming calls directly to the call queue associated with the menu number specified in this command. Once in the queue, a caller hears ringback if an agent is available or music on hold (MOH) if all agents are busy. If an greeting prompt for drop-through mode is configured, a caller hears the prompt before being sent to the queue as described.

The menu option number is an aa-hunt number that is associated with an ephone hunt group using the **param aa-hunt** command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with two AAs, both in drop-through mode. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. Callers to (800) 555-0121 drop directly through to the hunt group with the pilot number 5072 after hearing the greeting prompt in the audio file named en_dto_welcome.au. (The prefix en is specified in the **paramspace language** command and is automatically added to the filename provided in the **param drop-through-prompt** command.) Callers to (800) 555-0123 drop directly through to the hunt group with the pilot number 5071 without hearing any greeting.

```
dial-peer voice 1010 pots
 service acdaa
 port 1/1/0
 incoming called-number 8005550121
```

```
dial-peer voice 1020 pots
 service aa-bcd
 port 1/1/1
 incoming called-number 8005550123
.
.
.
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
  param number-of-hunt-grps 2
  param queue-len 10
!
 service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550121
  param service-name callq
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 20
  param number-of-hunt-grps 1
  param drop-through-prompt _bacd_dto_welcome.au
  param drop-through-option 2
  param second-greeting-time 45
  param handoff-string acdaa
  param max-time-call-retry 360
!
 service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550123
  param service-name callq
  param second-greeting-time 60
  param max-time-call-retry 180
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 5
  param handoff-string aa-bcd
  param drop-through-option 1
  param number-of-hunt-grps 1
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param ea-password

To create a password for accessing the extension assigner application, use the **param ea-password** command in application-parameter configuration mode.

param ea-password *password*

Syntax Description

password	Numeric string to be used as password for the extension assigner application. Password string must be 2 to 10 characters long and can contain numbers 0 to 9.

Command Default

No password is created.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command creates a password for using the extension assigner application.

If this command is not configured, you cannot use the extension assigner application.

Note There is no **no** form of this command. To change or remove the password for the extension assigner application, remove the service using the **no** form of the **service** command in application configuration mode.

Examples

The following example shows that a password (1234) is configured for the extension assigner application:

```
application
 service EA flash:ea/app-cme-ea-2.0.0.0.tcl
  paramspace english index 0
  paramspace english language en
  param ea-password 1234
  paramspace english location flash:ea/

paramspace english prefix en
```

Related Commands

	Description
application	Enters application configuration mode.

	Description
service	Loads and configures a specific, standalone application on a dial peer.

param handoff-string

To specify the name of a Cisco Unified CME B-ACD auto-attendant (AA) to be passed to the call-queue script, use the **param handoff-string** command in application-parameter configuration mode. To disable the handoff string, use the no form of this command.

param handoff-string *aa-service-name*
no param drop-through-prompt *aa-service-name*

Syntax Description

aa-service-name	Service name that was assigned to the AA script with the **service** command.

Command Default

No string is designated to be passed to the call-queue service.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice handoff-string** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

The handoff string is used only when the call-queue script starts for the first time or restarts after a failure.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number drop through to the ephone hunt group that has a pilot number of 5071 after hearing the initial prompt from the file en_dt_prompt.au. The AA name, aa is passed to the call-queue service in the **param handoff-string** command.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100
ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
```

```
paramspace english location tftp://192.168.254.254/user1/prompts/
paramspace english index 0
paramspace english language en
param aa-pilot 8005550100
param number-of-hunt-groups 1
param service-name callq
param handoff-string aa
param second-greeting-time 60
param drop-through-option 1
param drop-through-prompt _dt_prompt.au
param call-retry-timer 15
param max-time-call-retry 700
param voice-mail 5000
param max-time-vm-retry 2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param max-extension-length

To specify the maximum number of digits callers can dial when they choose the dial-by-extension option from the Cisco Unified CME B-ACD service, use the **param max-extension-length** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param max-extension-length *number*
no param max-extension-length *number*

Syntax Description

number	Number of digits. The lower limit is 0; there is no upper limit. The default is 5.

Command Default

The default number of digits callers can dial using the dial-by-extension option is 5.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice max-extension-length** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

Use this command to restrict the number of digits that callers can dial when using the dial-by-extension option.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100
ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
```

```
paramspace english index 0
paramspace english language en
param aa-pilot 8005550100
param welcome-prompt _aa_welcome.au
param number-of-hunt-groups 1
param dial-by-extension-option 2
param max-extension-length 4
param service-name callq
param handoff-string aa
param second-greeting-time 60
param call-retry-timer 15
param max-time-call-retry 700
param voice-mail 5000
param max-time-vm-retry 2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param max-time-call-retry

To specify the maximum length of time for which calls to the Cisco Unified CME B-ACD service can stay in a call queue, use the **param max-time-call-retry command in** application-parameter configuration mode. To return to the default, use the **no** form of this command.

param max-time-call-retry *seconds*
no param max-time-call-retry

Syntax Description

seconds	Maximum length of time that the call-queue service can keep redialing a hunt group pilot number, in seconds. Range: 20 to 3600. Default: 600.

Command Default

A call in a B-ACD call queue continues to try to connect to a hunt group for 600 seconds.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice max-time-call-retry** command.
12.4(20)YA	Cisco Unified CME 7.0(1)	The minimum value of the *seconds* argument was increased from 0 to 20.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service. Configure this command under the **service** command for an AA service.

A call to a Cisco Unified CME B-ACD service is put into a call queue if the hunt group that the call tried to reach has no phones available to take the call because they are all busy. While the call is in the queue, a second greeting is played at intervals specified by the **param second-greeting-time** command. From the queue, the call makes retries to connect at intervals specified by the **param call-retry-timer** command until the maximum amount of time to be spent in the queue expires. The maximum amount of time is set by the **param max-time-call-retry** command. After the maximum amount of time expires, the call is routed to the alternate destination specified in the **param voice-mail** command. If the alternate destination number is busy, the call makes the number of retries to connect specified in the **param max-time-vm-retry** command. If the call is unable to connect to the alternate destination after the number of retries that has been specified, it is disconnected.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the "Setting Up Call-Queue and AA Services" section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is “Please continue to hold. An agent will be with you shortly.” Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is disconnected.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100

ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

Command	Description
application	Enters application configuration mode.
call application voice load	Reloads the selected voice application script after it is modified.
param call-retry-timer	Specifies the time interval before each attempt to retry to connect a call to an ephone hunt group in a Cisco Unified CME B-ACD service.
param max-time-vm-retry	Specifies the maximum number of times that calls in a Cisco Unified CME B-ACD call queue can attempt to connect to the alternate destination number.

Command	Description
param second-greeting-time	Sets the length of the intervals between replays of the second greeting to calls waiting in hunt group call queues that are part of a Cisco Unified CME B-ACD service.
param voice-mail	Sets an alternate destination number to which to route calls that cannot be connected to a hunt group that is part of a Cisco Unified CME B-ACD service.
service	Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.

param max-time-vm-retry

To specify the maximum number of times that calls in a Cisco Unified CME B-ACD call queue can attempt to connect to the alternate destination number, use the **param max-time-vm-retry** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param max-time-vm-retry *number*
no param max-time-vm-retry *number*

Syntax Description

number	Number of times that the alternate destination number is redialed by the call-queue service. Range is from 1 to 3. Default is 1.

Command Default

A call in a B-ACD call queue tries to connect to an alternate destination number 1 time.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice max-time-vm-retry** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

A call to a Cisco Unified CME B-ACD service is put into a call queue if the hunt group that the call tried to reach has no phones available to take the call because they are all busy. While the call is in the queue, a second greeting is played at intervals specified by the **param second-greeting-time** command. From the queue, the call makes retries to connect at intervals specified by the **param call-retry-timer** command until the maximum amount of time to be spent in the queue expires. The maximum amount of time is set by the **param max-time-call-retry** command. After the maximum amount of time expires, the call is routed to the alternate destination specified in the **param voice-mail** command. If the alternate destination number is busy, the call makes the number of retries to connect specified in the **param max-time-vm-retry command.** If the call is unable to connect to the alternate destination after the number of retries that has been specified, it is disconnected.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is “Please continue to hold. An agent will be with you shortly.” Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is now disconnected.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100
ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param menu-timeout

To set the number of times the AA service will loop the menu prompt before connecting the caller to an operator if the caller does not select a menu option, use the **param menu-timeout** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param menu-timeout *number*
no param menu-timeout

Syntax Description

number	Times to replay menu prompt before connecting a caller to an operator. Range: 0 to 10. Default: 4.

Command Default

Auto-attendant service replays menu prompt 4 times before connecting the caller to an operator.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced.
12.4(22)YA	Cisco Unified CME 7.0(1)	The minimum value of the *number* argument was decreased from 1 to 0.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command is used with Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service.

If a caller does not select a menu option before the timeout set with this command expires, the call is transferred to the operator hunt group. The operator hunt-group is the hunt group with the highest aa-hunt number set with the **param aa-hunt** command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the "Setting Up Call-Queue and AA Services" section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

Examples

The following example shows the menu timeout set to 5 replays for the AA application called order1-aa:

```
application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/app-b-acd-aa-2.1.2.3.tcl
  paramspace english index 1
  param menu-timeout 5
  param handoff-string acme-aa1
  param dial-by-extension-option 2
  paramspace english language en
  param max-time-vm-retry 2
  param max-extension-length 4
  param aa-pilot 8005550100
  paramspace english location flash:/bacd/
  param second-greeting-time 60
```

```
param welcome-prompt _aa_welcome.au
param number-of-hunt-groups 1
param call-retry-timer 15
param max-time-call-retry 700
param voice-mail 5000
param service-name callq
```

Related Commands

Command	Description
application	Enters application configuration mode.
call application voice load	Reloads the selected voice application script after it is modified.
param aa-hunt	Declares a Cisco Unified CME B-ACD menu number and associates it with the pilot number of an ephone hunt group.
service	Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.

param number-of-hunt-grps

To specify the number of hunt groups used with a Cisco Unified CME B-ACD call-queue or AA service, use the **param number-of-hunt-grps** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param number-of-hunt-grps *number*
no param number-of-hunt-grps *number*

Syntax Description

number	Number of ephone hunt groups used by the service. Range is 1 to 10 for the call-queue service and 1 to 3 for an automated attendant (AA) service.

Command Default

This parameter is not set.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice number-of-hunt-grps** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured both under the **service** command for the call-queue service and under the **service** command for an AA service.

The number of hunt groups specified for the call-queue service is the total of the number of hunt groups used with all the AAs with which it is associated. For example, if a B-ACD has three AAs, each with two hunt groups, the number of hunt groups for each AA is two and the number of hunt groups for the call-queue service is six.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

A call-queue service called CQ is set up to work with two AA services. CQ lists 4 as the number of hunt groups it uses. AA1 is associated with 3 hunt groups, and its callers hear the following prompt: “Press 1 for sales, press 2 for service, press 0 for operator.” AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it.

```
application
 service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
  param aa-hunt4 4001
  param number-of-hunt-grps 4
  param queue-len 10
 service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
```

```
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550111
 param number-of-hunt-groups 3
 param service-name CQ
 param welcome-prompt _bacd_welcome.au
 param handoff-string AA1
service AA2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550122
 param number-of-hunt-groups 1
 param service-name CQ
 param drop-through-option 4
 param handoff-string AA2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param queue-exit-extension

To assign an extension number to a call-queue exit option, use the **param queue-exit-extension** command in application-parameter configuration mode. To remove an exit option, use the **no** form of this command.

param queue-exit-extension *option-number extension-number*
no param queue-exit-extension *option-number*

Syntax Description

option-number	Number of the call-queue exit option. Range: 1 to 3. There is no default.
extension-number	Extension number associated with the exit option.

Command Default

Call-queue exit option is not defined.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YA	Cisco Unified CME 7.0(1)	This command was introduced.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service.

Use this command together with the **param queue-exit-option** command to enable callers to select from up to three different options to exit from a call queue. The *option-number* argument in this command corresponds to the *option-number* argument in the **param queue-exit-option** command.

You can record a customized second greeting that offers callers up to three options to exit from the call queue. For example, you might record a message that says, "To leave a message, press 6; to hear more options, press 7; to speak to an operator, press 8."

This second greeting is stored in the audio file named en_bacd_allagentsbusy.au. You can record over the default message in this file, provided you do not change the name of the file.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the "Setting Up Call-Queue and AA Services" section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

Examples

The following example shows that the acme-aa1 application has three exit options defined for its call-queue service:

```
application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/app-b-acd-aa-2.1.2.3.tcl
  param dial-by-extension-option 7
  param handoff-string acme-aa1
  paramspace english index 1
  param queue-exit-option2 7
  param max-time-vm-retry 2
```

```
paramspace english language en
param aa-pilot 801
param max-extension-length 4
param queue-overflow-extension 101
param queue-exit-extension2 101
param second-greeting-time 20
param queue-exit-option1 6
paramspace english location flash:/bacd/
param send-account true
param call-retry-timer 20
param queue-exit-option3 8
param voice-mail 444
param max-time-call-retry 60
param service-name sf-queue
param queue-exit-extension1 202
param number-of-hunt-grps 1
param drop-through-option 1
param queue-exit-extension3 444
```

Related Commands

Command	Description
application	Enters application configuration mode.
call application voice load	Reloads the selected voice application script after it is modified.
param queue-exit-option	Assigns a menu number to a call-queue exit option.
service	Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.

param queue-exit-option

To assign a menu number to a call-queue exit option, use the **param queue-exit-option** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param queue-exit-option *option-number menu-number*
no param queue-exit-option *option-number*

Syntax Description

option-number	Number of the call-queue exit option. Range: 1 to 3. There is no default.
menu-number	Menu option number associated with the exit option.

Command Default

Call-queue exit option is not assigned.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YA	Cisco Unified CME 7.0(1)	This command was introduced.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service.

Use this command together with the **param queue-exit-extension** command to enable callers to select from up to three different options to exit from a call queue. The *option-number* argument in this command corresponds to the *option-number* argument in the **param queue-exit-extension** command.

You can record a customized second greeting that offers callers up to three options to exit from the call queue. For example, you might record a message that says, "To leave a message, press 6; to hear more options, press 7; to speak to an operator, press 8."

This second greeting is stored in the audio file named en_bacd_allagentsbusy.au. You can record over the default message in this file, provided you do not change the name of the file.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the "Setting Up Call-Queue and AA Services" section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

Examples

The following example shows that the acme-aa1 application has three exit options defined for its call-queue service:

```
application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/app-b-acd-aa-2.1.2.3.tcl
  param dial-by-extension-option 7
  param handoff-string acme-aa1
  paramspace english index 1
  param queue-exit-option2 7
  param max-time-vm-retry 2
```

```
paramspace english language en
param aa-pilot 801
param max-extension-length 4
param queue-overflow-extension 101
param queue-exit-extension2 101
param second-greeting-time 20
param queue-exit-option1 6
paramspace english location flash:/bacd/
param send-account true
param call-retry-timer 20
param queue-exit-option3 8
param voice-mail 444
param max-time-call-retry 60
param service-name sf-queue
param queue-exit-extension1 202
param number-of-hunt-grps 1
param drop-through-option 1
param queue-exit-extension3 444
```

Related Commands

Command	Description
application	Enters application configuration mode.
call application voice load	Reloads the selected voice application script after it is modified.
param queue-exit-extension	Assigns an extension number to a call-queue exit option.
service	Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.

param queue-len

To specify the number of calls that can be held in each call queue in a Cisco Unified CME B-ACD service, use the **param queue-len** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param queue-len *number*
no param queue-len *number*

Syntax Description

number	Number of calls that can be held in a call queue. Range is 1 to 30. Default is 10.

Command Default

The default queue length is 10.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice queue-len command.**

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for a call-queue service.

This command specifies the maximum number of calls that can be held in a call queue for a hunt group used with B-ACD when all of the hunt group member phones are busy.

Note that having calls in queue keeps PSTN ports occupied for a longer time, and you may want to plan for more ports if you have longer queues. The maximum number of calls allowed in the queues of ephone hunt groups must be based on the number of ports or trunks available. For example, if you have 20 foreign exchange office (FXO) ports and two ephone hunt groups, you can configure a maximum of ten calls per ephone hunt-group queue using the **param queue-len 10** command. You can use the same configuration if you have a single T1 trunk, which supports 23 channels.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

A call-queue service called CQ is set up to work with two AA services. CQ lists four as the number of hunt groups it uses. AA1 is associated with three hunt groups, and its callers hear the following prompt: "Press 1 for sales, press 2 for service, press 0 for operator." AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it. Up to 12 calls can be held in the call queue for each hunt group if all the phones in the hunt group are busy.

```
application
 service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
```

```
 param aa-hunt4 4001
 param number-of-hunt-grps 4
 param queue-len 12
service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550111
 param number-of-hunt-groups 3
 param service-name CQ
 param welcome-prompt _bacd_welcome.au
 param handoff-string AA1
service AA2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550122
 param number-of-hunt-groups 1
 param service-name CQ
 param drop-through-option 4
 param handoff-string AA2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param queue-manager-debugs

To enable the collection of call-queue debug information in a Cisco Unified CME B-ACD service, use the **param queue-manager-debugs** command in application-parameter configuration mode. To remove the setting, use the **no** form of this command with the keyword that was previously used.

param queue-manager-debugs [{**0** | **1**}]
no param queue-manager-debugs [{**0** | **1**}]

Syntax Description

0	Disables collection of call-queue debug information.
1	Enables collection of call-queue debug information

Command Default

Collection of debug information is disabled.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice queue-manager-debugs command.**

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for the call-queue service.

This command enables the collection of data regarding call queue activity. It is used in conjunction with the **debug voip ivr script command.** Both commands must be enabled at the same time.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

A call-queue service called CQ is set up to work with two AA services. CQ lists four as the number of hunt groups it uses. AA1 is associated with three hunt groups, and its callers hear the following prompt: “Press 1 for sales, press 2 for service, press 0 for operator.” AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it. Up to ten calls can be held in the call queue for each hunt group if all the phones in the hunt group are busy with other calls. Call-queue debugging is enabled.

```
application
 service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
  param aa-hunt4 4001
  param number-of-hunt-grps 4
  param queue-len 10
 service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
```

```
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550111
 param number-of-hunt-groups 3
 param service-name CQ
 param welcome-prompt _bacd_welcome.au
 param handoff-string AA1
service AA2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550122
 param number-of-hunt-groups 1
 param service-name CQ
 param drop-through-option 4
 param handoff-string AA2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param queue-overflow-extension

To set the extension number to route calls to when the call queue for the auto-attendant service is full, use the **param queue-overflow-extension** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param queue-overflow-extension *extension-number*
no param queue-overflow-extension

Syntax Description

extension-number	Extension number to which the auto-attendant service forwards calls when the call queue is full.

Command Default

No overflow extension is defined. Calls disconnect if the queue becomes full.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YA	Cisco Unified CME 7.0(1)	This command was introduced.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service.

This command specifies the extension number where calls are sent when the number of calls waiting in a B-ACD call queue exceeds the number set with the **param queue-len** command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For configuration information, see the "Setting Up Call-Queue and AA Services" section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

Examples

The following example shows that the AA application named acme-aa1 uses the call-queue service named CQ. When the number of calls in the queue exceeds 12, new calls that cannot be answered by an agent are sent to extension 5100.

```
application
  service CQ tftp://192.168.254.254/acme/bacd/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
  param aa-hunt4 4001
  param number-of-hunt-grps 4
  param queue-len 12
  !
!
 application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/app-b-acd-aa-2.1.2.3.tcl
  paramspace english index 1
```

```
param handoff-string acme-aa1
param dial-by-extension-option 2
paramspace english language en
param aa-pilot 8005550100
param queue-overflow-extension 5100
paramspace english location flash:/bacd/
param welcome-prompt _aa_welcome.au
param number-of-hunt-groups 1
param voice-mail 5000
param service-name CQ
```

Related Commands

Command	Description
application	Enters application configuration mode.
call application voice load	Reloads the selected voice application script after it is modified.
param queue-len	Specifies the number of calls that can be held in each call queue in a Cisco Unified CME B-ACD service.
service	Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.

param secondary-prefix

To set a prefix to add to digits coming from the PSTN Central Office (CO) to route calls from a secondary Cisco Unified CME router to a primary Cisco Unified CME router when using the Direct Inward Dial (DID) Digit Translation Service, use the **param secondary-prefix** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param secondary-prefix *secondary-prefix*
no param secondary-prefix *secondary-prefix*

Syntax Description

secondary-prefix	Prefix to add to digits in order to route calls to the primary Cisco Unified CME router. Range is from 0 to 99.

Command Default

No prefix is defined.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice secondary-prefix** command.
12.4(9)T	Cisco Unified CME 4.0	This command replaced the **call application voice secondary-prefix** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service, which provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

When calls are received by a secondary Cisco Unified CME router, they are routed to the primary router by configuring an H.323 VoIP dial peer and matching the destination pattern for that dial peer. The DID prefix that was configured for use with the DID script is appended to the incoming DID digits first. The secondary prefix is appended next. For example, if the incoming DID digits are 25, the DID prefix is 3, and the secondary prefix is 7, the transformed number will be 7325. The transformed number matches a VoIP dial peer, which uses the **forward-digits** command to send only the three relevant digits, the extension number, to the primary router.

See the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example configures a Basic DID application on the Cisco Unified CME router. It sets a prefix of 5 to apply to the digits coming from the CO in order to construct a valid extension number. Then the secondary prefix (4) is appended. If the incoming DID digits are 25, the DID prefix is 5, and the secondary prefix is 4, then the transformed number is 4525. The transformed number matches VoIP dial peer 1000. The VoIP dial peer sends calls to the primary Cisco Unified CME router using the IP address that is entered in the session target command. The dial peer uses the **forward-digits** command to send the extension number, 525, to the primary Cisco Unified CME router.

```
dial-peer voice 1000 voip
 destination-pattern 45..
 session target ipv4:10.1.1.1
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 forward-digits 3
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
  param co-did-max 79
  param store-did-min 00
  param store-did-max 79
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param second-greeting-time

To set the length of the intervals between playouts of the second greeting to calls waiting in hunt group call queues that are part of a Cisco Unified CME B-ACD service, use the **param second-greeting-time** command in application-parameter configuration mode. To return to the default, use the **no** form of this command

param second-greeting-time *seconds*
no param max-time-vm-retry *seconds*

Syntax Description

seconds	Length of time intervals between playouts of the second greeting to calls in a B-ACD call queue, in seconds. Range is from 30 to 120. Default is 60.

Command Default

The second greeting is played out every 60 seconds to calls in B-ACD call queues.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice second-greeting-time command.**

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

A call to a Cisco Unified CME B-ACD service is put into a call queue if the hunt group that the call tried to reach has no phones available to take the call because they are all busy. While the call is in the queue, a second greeting is played at intervals specified by the **param second-greeting-time** command. From the queue, the call retries to connect to the hunt group at intervals specified by the **param call-retry-timer** command until the maximum amount of time to be spent in the queue expires. The maximum amount of time is set by the **param max-time-call-retry** command. After the maximum amount of time expires, the call is routed to the alternate destination specified in the **param voice-mail** command. If the alternate destination number is busy, the call makes the number of retries to connect specified in the **param max-time-vm-retry command.** If the call is unable to connect to the alternate destination after the number of retries that has been specified, it is disconnected.

The second greeting is stored in the audio file named en_bacd_allagentsbusy.au. You can rerecord over the default message that is provided in the file, but you cannot change the name of the file.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is “Please continue to hold. An agent will be with you shortly." Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is now disconnected.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100
ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param send-account true

To generate call detail records (CDRs) for calls that are handled by the Cisco Unified CME B-ACD service, use the **param send-account** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param send-account true
no param send-account true

Syntax Description

This command has no arguments or keywords.

Command Default

CDRs are not generated.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command captures CDRs in RADIUS format for calls handled by the Cisco Unified CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) service. The call record includes the name of the AA service, hunt group pilot-number, and globally unique identifier (GUID).

For configuration information, see the "Setting Up Call-Queue and AA Services" section in the *Cisco Unified CME B-ACD and Tcl Call-Handling Applications* document.

For information on enabling RADIUS accounting, see the CDR Accounting for Cisco IOS Voice Gateways guide.

Examples

The following example shows that calls using the acme-aa1 service generate a call detail record:

```
application
  service acme-aa1 tftp://192.168.254.254/acme/bacd/
app-b-acd-aa-2.1.2.3.tcl
  paramspace english index 1
  param handoff-string acme-aa1
  param dial-by-extension-option 2
  paramspace english language en
  param aa-pilot 8005550100
  paramspace english location flash:/bacd/
  param welcome-prompt _aa_welcome.au
  param send-account true
  param number-of-hunt-groups 1
  param voice-mail 5000
  param service-name callq
```

Related Commands

Command	Description
application	Enters application configuration mode.

Command	Description
call application voice load	Reloads the selected voice application script after it is modified.
gw-accounting aaa	Enables the gateway to send accounting CDRs to the RADIUS server using VSAs (attribute 26).
service	Enters application-parameter configuration mode to specify the name of the application and the location of the Tcl script to load.

param service-name

To specify a Cisco Unified CME B-ACD call-queue service to use with an automated attendant (AA) service, use the **param service-name** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param service-name *queue-service-name*
no param service-name *queue-service-name*

Syntax Description

queue-service-name	Name that was assigned to the B-ACD call-queue service with the **service** command.

Command Default

No call-queue service is specified.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice service-name command.**

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

A call-queue service called CQ is set up to work with two AA services. CQ lists four as the number of hunt groups it uses. AA1 is associated with three hunt groups, and its callers hear the following prompt: "Press 1 for sales, press 2 for service, press 0 for operator." AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it. Up to ten calls can be held in the call queue for each hunt group if all the phones in the hunt group are busy with other calls. Call-queue debugging is enabled.

```
application
 service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
  param aa-hunt4 4001
  param number-of-hunt-grps 4
  param queue-len 10
 service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550111
  param number-of-hunt-groups 3
  param service-name CQ
  param welcome-prompt _bacd_welcome.au
```

```
 param handoff-string AA1
service AA2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550122
 param number-of-hunt-groups 1
 param service-name CQ
 param drop-through-option 4
 param handoff-string AA2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param store-did-max

To set the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan used with the Direct Inward Dial (DID) Digit Translation Service, use the **param store-did-max** command in global configuration mode. To disable this option, use the **no** form of this command.

param store-did-max *max-store-value*
no param store-did-max *max-store-value*

Syntax Description

max-store-value	Maximum value of digits in the Cisco Unified CME dial plan. The digit string can be any length, but the string length must be the same in the **param co-did-max**, **param co-did-min**, **param store-did-max**, and **param store-did-min** commands.

Command Default

No maximum value is defined for the range of digits in the dial plan.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice store-did-max** command.
12.4(9)T	Cisco Unified CME 4.0	This command replaced the **call application voice store-did-max** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command defines the upper limit of the range of digits in the site dial plan for the Cisco Unified CallManager Express (Cisco Unified CME) Direct Inward Dial Digit Translation Service, which provides number translation for DID calls when the DID digits provided by the PSTN Central Office (CO) do not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers. A prompt is played and the calls are disconnected.

Examples

The following example configures Direct Inward Dial Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the range of digits in the Cisco Unified CME extension dial plan. Notice that the length of the digit string is the same (2 digits) for all related commands.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
```

```
param did-prefix 5
param co-did-min 00
param co-did-max 79
param store-did-min 00
param store-did-max 79
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.
param co-did-max	Sets the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the Direct Inward Dial Digit Translation Service.
param co-did-min	Sets the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the Direct Inward Dial Digit Translation Service.
param store-did-min	Sets the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the Direct Inward Dial Digit Translation Service.

param store-did-min

To set the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan used with the Direct Inward Dial (DID) Digit Translation Service, use the **param store-did-min** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param store-did-min *min-store-value*
no param store-did-min *min-store-value*

Syntax Description

min-store-value	Minimum value of digits in the Cisco Unified CME dial plan. The digit string can be any length, but the string length must be the same in the **param co-did-max**, **param co-did-min**, **param store-did-max**, and **param store-did-min** commands.

Command Default

No minimum value is defined for the range of digits in the dial plan.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice store-did-min command.**
12.4(9)T	Cisco Unified CME 4.0	This command replaced the **call application voice store-did-min** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command defines the lower limit of the range of digits in the site dial plan when it is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service. This service provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

Examples

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the range of digits in the Cisco Unified CME extension dial plan.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
```

```
param co-did-max 79
param store-did-min 00
param store-did-max 79
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.
param co-did-max	Sets the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.
param co-did-min	Sets the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.
param store-did-max	Sets the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.

param voice-mail

To set an alternate destination number to which to route calls that cannot be connected to a hunt group that is part of a Cisco Unified CME B-ACD service, use the **param voice-mail** command in application-parameter configuration mode. To return to the default, use the **no** form of this command

param voice-mail *number*
no param voice-mail *number*

Syntax Description

number	Extension number to which to route calls. The number must be associated with a dial peer that is reachable by the Cisco Unified CME system.

Command Default

No alternate destination number is set.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice voice-mail command.**

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

Calls are diverted to an alternate destination only when one of the following criteria is met:

- The hunt group to which the call has been transferred is unavailable because all members are logged out.
- The call-queue maximum retry timer has expired.

The alternate destination can be any number at which you can assure call coverage, such as a voice-mail number, a permanently staffed number, or a number that rings an overhead night bell. Once a call is diverted to an alternate destination, it is no longer controlled by the B-ACD service. This parameter is set with the **param voice-mail** command.

If you send calls to a voice-mail system as an alternate destination, be sure to set up the voice-mail system as specified in the documentation for the system.

If you specify a number for an alternate destination, the number must be associated with a dial peer that is reachable by the Cisco Unified CME system.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information about B-ACD, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is "Please continue to hold. An agent will be with you shortly." Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000, which is the alternate destination. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is disconnected.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100
ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param welcome-prompt

To specify an audio file containing a prompt to be played as a welcome for callers to an automated attendant (AA) that is part of a Cisco Unified CME B-ACD service, use the **param welcome-prompt** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param welcome-prompt *audio-filename*
no param welcome-prompt *audio-filename*

Syntax Description

audio-filename	Identifier part of name of the audio file that contains the welcome greeting to be played when callers first reach the Cisco Unified CME B-ACD service. This name does not include the language prefix and it must begin with an underscore. Default is _bacd_welcome.au.

Command Default

The audio file named en_bacd_welcome.au is used as a welcome prompt.

Command Modes

Application-parameter configuration (config-app-param)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice voice-mail command.**

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

Each AA service that is used with the Cisco Unified CME B-ACD service needs a welcome greeting to tell callers the destination they have reached and, sometimes, the options that they have. The en_bacd_welcome.au audio file is used by default. It announces “Thank you for calling,” and includes a two-second pause after the message. The filename of the welcome prompt audio file has two parts: a two-letter prefix that denotes a language code specified in the **paramspace language** command, and the identifying part that indicates the purpose of the file. In the default welcome prompt audio file, the prefix is en and the identifying part is _bacd_welcome.au. Note that the identifying part starts with an underscore.

If your Cisco Unified CME B-ACD service uses a single AA application, you can record a custom welcome greeting in the audio file named en_welcome_prompt.au and record instructions about menu choices in the audio file named en_bacd_options_menu.au.

If your Cisco Unified CME B-ACD service uses multiple AA applications, you will need separate greetings and menu options for each AA. Use the following guidelines:

- Record a separate welcome prompt for each AA application, using a different name for the audio file for each welcome prompt. For example, en_welcome_aa1.au and en_welcome_aa2.au. The welcome prompts that you record in these files should include both the greeting and the instructions about menu options.
- Record silence in the audio file en_bacd_options_menu.au. A minimum of one second of silence must be recorded. Note that you cannot change the identifier part of the name of this audio file.

For any Cisco Unified CME B-ACD configuration changes to take effect, you must reload the scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for two AA applications, called aa1 and aa2, and a call-queue application called queue. The direct-dial numbers to reach the AA services are (800) 555-0100 for aa1 and (800) 555-0110 for aa2. Callers to aa1 can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits. Callers to aa2 can press 2 to dial an extension number of 4 or fewer digits or press 3 to be connected to the ephone hunt group with the pilot number 5073. Both AAs share an operator hunt group, which is menu option 4.

The welcome prompt for aa1 is “Thank you for calling the Sales department. Press 1 to place an order. Press 2 if you know the extension of the party you want, or press 0 to speak to an operator.” The filename of the audio file that contains this welcome prompt is en_aa1_welcome.au.

The welcome prompt for aa2 is “Thank you for calling the Service department. Press 2 if you know the extension of the party you want. Press 3 to speak to a service technician or press 0 to speak to an operator.” The filename of the audio file that contains this welcome prompt is en_aa2_welcome.au.

```
dial-peer voice 1000 pots
 service aa1
 port 1/1/0
 incoming called-number 8005550100
dial-peer voice 1100 pots
 service aa2
 port 1/1/1
 incoming called-number 8005550110
ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
ephone-hunt 11 sequential
 pilot 5073
 list 5021, 5022, 5023, 5024, 5025
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt3 5073
  param aa-hunt4 6000
  param number-of-hunt-grps 3
  param queue-len 10
!
 service aa1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa1_welcome.au
  param number-of-hunt-groups 2
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa1
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
 service aa2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
```

```
param aa-pilot 8005550110
param welcome-prompt _aa2_welcome.au
param number-of-hunt-groups 2
param dial-by-extension-option 2
param max-extension-length 4
param service-name callq
param handoff-string aa2
param second-greeting-time 60
param call-retry-timer 15
param max-time-call-retry 700
param voice-mail 5000
param max-time-vm-retry 2
```

Related Commands

	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

paramspace callsetup after-hours-exempt

To specify that an individual dial peer does not have any of its calls blocked by the Cisco router even though call blocking has been enabled, use the **paramspace callsetup after-hours-exempt** command in dial-peer configuration mode. To return to the default, use the no form of this command.

paramspace callsetup after-hours-exempt {**true** | **false**}
no paramspace callsetup after-hours-exempt

Syntax Description

true	Dial peer is exempt from call-blocking configuration.
false	Dial peer is subject to call-blocking configuration. This is default.

Command Default

All dial peers are subject to call-blocking configuration.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Cisco IOS Release	Cisco Products	Modification
12.4(4)T	Cisco CME 3.4 Cisco SRST 3.4	This command was introduced.

Usage Guidelines

This command is intended to allow H.323 and SIP trunk calls to utilize the voice gateway in spite of the the after-hours configuration in Cisco Unified CME or Cisco Unified SRST.

A Cisco voice gateway (session application) accesses the after-hours call-blocking configuration set by Cisco Unified CME or Cisco Unified SRST and blocks *all* SCCP, SIP, H.323, and POTS calls that go through the Cisco router regardless of whether the call is from a phone controlled by the Cisco router or from a phone controlled by some other call control application, such as Cisco Unified CallManager.

To disable the After Hours Call Blocking feature for incoming calls from phones other than those registered to a Cisco Unified CME or Cisco Unified SRST router, use this command to exempt an individual H.323, SIP, or POTS dial peer from the call blocking configuration.

To disable the After Hours Call Blocking feature for an individual IP phone registered in Cisco Unified CME or Cisco Unified SRST:

- In Cisco CME 3.4 and later, disable the After Hours Call Blocking feature for a directory number on a SIP phone by using the **after-hour exempt** command in voice register pool or voice register dn configuration mode.
- In Cisco CME 3.0 and later, disable the After Hours Call Blocking feature for an individual SCCP phone by using the **after-hour exempt** command in ephone or ephone-template configuration mode.
- In Cisco SIP SRST 3.4 and later, disable the After Hours Call Blocking feature for SIP phones in a voice register pool by using the **after-hour exempt** command in voice register pool configuration mode.
- In Cisco SRST, you cannot create an exemption for an individual phone from the call-blocking configuration.

Examples

The following example shows how to set the After Hours Call Blocking feature in Cisco Unified CME, and how to configure a particular dial peer (255) so that outgoing calls through this dial peer are exempt from this after-hours call blocking configuration:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 9011
Router(config-telephony)# exit
Router(config)# dial-peer voice 255 voip
Router(config-dial-peer)# paramspace callsetup after-hours-exempt true
```

Related Commands

	Description
after-hour exempt	Specifies that a SCCP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hour exempt (voice register dn)	Specifies that an individual SIP IP phone or a phone extension on a SIP IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hour exempt (voice register pool)	Specifies that an individual SIP IP phone or phones in a voice register pool does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.

park reservation-group

To assign a call-park reservation group to a phone, use the **reservation-group** command in ephone, ephone-template, voice register pool, or voice register template configuration mode. To remove the group from the phone, use the **no** form of this command.

park reservation-group *group-number*
no park reservation-group

Syntax Description

group-number	Unique number that identifies the reservation group. String can contain up to 32 digits.

Command Default

Extension does not belong to any reservation group.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command allows you to assign ownership to call-park slots by using Park Reservation Groups. A phone configured with a park reservation group can retrieve calls only from park slots configured with the same park reservation group. A phone without a park reservation group can retrieve calls from any park slot without an assigned park reservation group.

To assign a reservation group to a park-slot extension, use the **park-slot reservation-group** command.

If you use a template to apply a command to a phone and you also use the same command in ephone or voice register pool configuration mode for the same phone, the value that you set in the phone configuration mode has priority.

Examples

The following example shows park reservation-group 1 is assigned to phone 3 (SCCP). When calls for the Pharmacy are parked at extension 8126, phone 3 can retrieve them:

```
ephone-dn  26
 number 8126
 park-slot reservation-group 1 timeout 15 limit 2 transfer 8100
 description park slot for Pharmacy
!
!
ephone  3
 park reservation-group 1
 mac-address 002D.264E.54FA
 type 7962
 button  1:3
```

The following example shows park reservation-group 1 is assigned to phone 120 (SIP). When calls for the Pharmacy are parked at extension 8126, phone 120 can retrieve them:

```
voice register pool  120
 park reservation-group 1
 id mac 0030.94C2.A22A
 type 7962
 number 1 dn 20
```

Related Commands

Command	Description
call-park system	Defines system parameters for the call-park feature.
park-slot	Creates an extension (call-park slot) at which calls can be temporarily held (parked).

park-slot

To create an extension (call-park slot) at which calls can be temporarily held (parked), use the **park-slot** command in ephone-dn configuration mode. To disable the extension, use the **no** form of this command.

park-slot [**directed**] [**reservation-group** *group-number*] [**reserved-for** *extension-number*] [[**timeout** *seconds* **limit** *count*] [**notify** *extension-number* [**only**]] [**recall**] [**transfer** *extension-number*] [**alternate** *extension-number*] [**retry** *seconds* **limit** *count*]]
no park-slot [**directed**] [**reservation-group** *group-number*] [**reserved-for** *extension-number*] [[**timeout** *seconds* **limit** *count*] [**notify** *extension-number* [**only**]] [**recall**] [**transfer** *extension-number*] [**alternate** *extension-number*] [**retry** *seconds* **limit** *count*]]

Syntax Description

directed	(Optional) Enables Directed Call Park for this extension.
reservation-group *group-number*	(Optional) Reserves this slot for phones configured with the same reservation group.
reserved-for *extension-number*	(Optional) Reserves this slot as a private park slot for the phone with the specified extension number as its primary line. All lines on that phone can use this park slot. **Note** This keyword is ignored if the **reservation-group** keyword is used.
timeout *seconds*	(Optional) Sets the call-park reminder timeout in seconds. Range: 0 to 65535. This reminder sends a 1-second ring to the IP phone that parked the call and displays a message on the LCD panel of all phones in the Cisco Unified CME system, indicating that a call is on hold. By default, the reminder ring is sent only to the phone that parked the call.
limit *count*	(Optional) Sets a limit on the number of reminder or retry timeouts. Range: 1 to 65535.
notify *extension-number*	(Optional) Sends a reminder ring to the specified extension in addition to the reminder ring that is sent to the phone that parked the call.
only	(Optional) Sends a reminder ring only to the extension specified with the **notify** keyword and does not send a reminder ring to the phone that parked the call. This option allows all reminder rings for parked calls to be sent to a receptionist's phone or an attendant's phone, for example.
recall	(Optional) Returns the call to the phone that parked it after the timeout expires.
transfer *extension-number*	(Optional) Returns the call to the specified extension after the timeout expires.
alternate *extension-number*	(Optional) Returns the call to this second target number if the recall or transfer target phone is in use on any of its extensions (ringing or connected).
retry *seconds*	(Optional) Sets the delay before another attempt to recall or transfer a parked call, in seconds. Range: 0 to 65535. Number of attempts is set by the **limit** keyword.

Command Default

No call-park slot is defined.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Release	Cisco Product	Modification
12.3(7)T	Cisco CME 3.1	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **reserved-for**, **recall**, **transfer**, **alternate**, and **retry** keywords were added.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(24)T.
12.4(22)YB	Cisco Unified CME 7.1	The **directed** and **reservation-group** keywords and support for SIP phones was added.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command creates a call-park slot that is a floating extension, or ephone-dn that is not bound to a physical phone, at which phone users can place calls on hold for later retrieval from the same phone or from another phone.

At least one call-park slot must be defined with this command before the Park soft key displays on IP phones in a Cisco Unified CME system.

Phone users park calls using the Park soft key. A phone user can then retrieve a call by dialing the extension number of the call-park slot. On SCCP phones, the phone user who parks the call can also retrieve the call by using the PickUp soft key and an asterisk (*). Other SCCP phone users can retrieve the call by using the PickUp soft key and dialing the extension number of the call-park slot.

Calls can also be transferred to a call-park slot using the Transfer key; a transfer to a call-park slot is always a blind transfer. Calls can also be forwarded to a call-park slot, and callers can directly dial call-park slots.

When a call that uses a G.711 codec is parked, the caller hears the music-on-hold (MOH) audio stream; otherwise, the caller hears the on-hold tone.

The **directed** keyword enables the extension as a park slot for Directed Call Park. To retrieve a call from a directed call-park slot, you must define the retrieval prefix with the **fac** command. The retrieval prefix is supported for both SCCP and SIP phones.

The **reservation-group** keyword allows you to assign ownership to call-park slots by using Park Reservation Groups. A park slot configured with a park reservation group can only be used by phones configured with the same park reservation group. A park slot without a park reservation group can be used by any phone not assigned to a park reservation group.

A reminder ring can be sent to the extension that parked the call by using the **timeout** keyword, which sets the interval length to wait before sending call-park reminder rings. The number of time-out intervals and reminder rings are configured with the **limit** keyword and argument. For example, a limit of 3 timeout intervals sends 2 reminder rings (interval 1, ring 1, interval 2, ring 2, interval 3). The **timeout** and **limit** keywords and arguments also set the maximum time that calls stay parked. For example, a timeout interval of 10 seconds and a limit of 5 timeout intervals (**park-slot timeout 10 limit 5**) will park calls for approximately 50 seconds.

If the **timeout** keyword is not used with this command, no reminder ring is sent to the extension that parked the call. If the **timeout** keyword is used, a reminder ring is sent only to the extension that parked the call unless the **notify** keyword is also used to specify an additional extension number to receive a reminder ring.

When an additional extension number is specified using the **notify** keyword, the phone user at that extension can retrieve a call from this slot by pressing the PickUp soft key and an asterisk (*).

Each call-park slot can hold one call at a time, so the number of simultaneous calls that can be parked is equal to the number of slots that have been created. The **reserved-for** keyword creates a call-park slot that is dedicated for use by one extension so that extension always has a slot available at which to park a call. With nonreserved slots, multiple call-park slots can be created with the same extension number so that all the calls that are parked for a particular group can be parked at a known extension number. For example, at a hardware store, calls for the plumbing department can be parked at extension 101, calls for lighting can be parked at 102, and so forth. Then, anyone in the plumbing department can pick up calls from extension 101. When multiple calls are parked at the same extension number, they are picked up in the order in which they were parked; that is, the call that has been parked the longest is the first call picked up from that extension number.

IP phone users park calls at their dedicated call-park slots using the Park soft key. Phone users can also transfer calls to dedicated call-park slots using the Transfer soft key and a standard or custom feature access code (FAC) for call park. On analog phones, users transfer calls to dedicated call-park slots using hookflash and a standard or custom FAC for call park. The standard FAC for call park is **6. Custom FACs are created using the **fac** command.

If a dedicated park slot is not found for an ephone-dn attempting to park a call, Cisco Unified CME uses the standard call-park procedure; that is, the system searches for a preferred park slot (one with an ephone-dn number that matches the last two digits of the ephone-dn attempting to park the call) and if none is found, uses any available call-park slot.

If a name has been specified for a call-park slot, that name is displayed instead of an extension number on a recall or transfer of the call.

A parked call can have the following dispositions after its timeouts expire:

- Recall—If you specify that a call should be recalled to the parking phone after the timeout interval expires, the call is always returned to the phone's primary extension number, regardless of which extension on the phone did the parking.
- Transfer—If you specify a transfer target, the call is transferred to the specified number after the timeout intervals expire instead of returning to the primary number of the phone that did the parking.
- Alternate—You can also specify an alternate target extension to which to transfer a parked call if the recall or transfer target is in use. *In use* is defined as either ringing or connected to a call. For example, a call is parked at the dedicated park slot for the phone with the primary extension of 2001. After the timeouts expire, the system attempts to recall the call to extension 2001, but that line is now connected to a different call. The system transfers the call to the alternate target that was specified when the park slot was defined.
- Disconnect—When a timeout limit is set and no other disposition has been specified, a call parked at a call-park slot is disconnected after the number of reminder timeouts are reached.

Examples

Basic Call Park

The following example shows a basic call-park slot at extension 1001. After a call is parked at this number, the system provides 10 reminder rings at intervals of 30 seconds to the extension that parked the call. Any phone can retrieve calls parked at this extension.

```
ephone-dn 45
 number 1001
 park-slot timeout 30 limit 10
```

Directed Call Park (Cisco Unified CME 4.4 and Later Versions)

The following example shows two call-park slots, extension 3110 and 3111, that can be used to park calls for the pharmacy using Directed Call Park.

```
ephone-dn  10
 number 3110
 park-slot directed
 description park-slot for Pharmacy
!
ephone-dn  11
 number 3111
 park-slot directed
 description park-slot for Pharmacy
```

Park Reservation Groups (Cisco Unified CME 4.4 and Later Versions)

The following example shows park reservation groups set up for two call-park slots. Extension 8126 is configured for group 1 and assigned to phones 3 and 4. Extension 8127 is configured for group 2 and assigned to phones 10 and 11. When calls for the Pharmacy are parked at extension 8126, only phones 3 and 4 can retrieve them.

```
ephone-dn  26
 number 8126
 park-slot reservation-group 1 timeout 15 limit 2 transfer 8100
 description park slot for Pharmacy
!
ephone-dn  27
 number 8127
 park-slot reservation-group 2 timeout 15 limit 2 transfer 8100
 description park slot for Auto
!
!
ephone  3
 park reservation-group 1
 mac-address 002D.264E.54FA
 type 7962
 button  1:3
!
ephone  4
 park reservation-group 1
 mac-address 0030.94C3.053E
 type 7962
 button  1:4
!
ephone  10
 park reservation-group 2
 mac-address 00E1.CB13.0395
 type 7960
 button  1:10
!
ephone  11
 park reservation-group 2
 mac-address 0016.9DEF.1A70
 type 7960
 button  1:11
```

Dedicated Park

The following example shows a dedicated call-park slot, 2558, that is reserved for the phone that has the primary extension of 2977. Both extension 2977 and 2976 are on the same phone, so they both can use this slot, which is reserved only for the extensions on that phone. After three timeout intervals of 60 seconds, a parked call is recalled to extension 2977. If extension 2977 is busy, the call is rerouted to extension 3754.

```
ephone-dn 24
 number 2977

ephone-dn 25
 number 2976

ephone-dn 27
 number 3754

ephone-dn 30
 number 2558
 name Park 2977
 park-slot reserved-for 2977 timeout 60 limit 3 recall alternate 3754

ephone 44
 button 1:24 2:25

ephone 45
 button 1:27
```

Related Commands

Command	Description
call-park system	Defines system parameters for the call-park feature.
fac	Enables standard FACs or creates custom FACs.
number	Associates a telephone or extension number with a directory number.
park reservation-group	Assigns a call-park reservation group to a phone.

password (auto-register)

To configure the mandatory password for automatic registration of SIP phones with the Cisco Unified CME system, use the **password** command in voice auto register configuration mode. This command is a sub-mode CLI of the command **auto-register** . To disable configuring password for auto registration of SIP phones, use the **no** form of this command.

password **[0|6]** *string*
no **password**

Syntax Description

password *string*	The mandatory word string that administrator provides for auto registration of phones on Unified CME.

Command Default

By default, this command is disabled.

Command Modes

voice auto register configuration (config-voice-auto-register)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption, based on Unified CME password policy.

Usage Guidelines

This command enables the administrator to configure the password credentials for SIP phones auto registering on Unified CME. It is mandatory that the password is configured before assigning the DN range.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]**. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Examples

The following example shows how to configure password for auto registration of SIP phones:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)# ?

VOICE auto register configuration commands:
auto-assign Define DN range for auto assignment
default Set a command to its defaults
exit Exit from voice register group configuration mode
no Negate a command or set its defaults
password Default password for auto-register phones
service-enable Enable SIP phone Auto-Registration
template Default template for auto-register phones

Router(config-voice-auto-register)#password ?
 WORD Password string
```

Related Commands

Command	Description
service-enable (auto-register)	Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.
auto-register	Enables automatic registration of SIP phones with the Cisco Unified CME system.
auto-assign (auto-register)	Configures the mandatory range of directory numbers for phones auto registering on Unified CME.
template (auto-register)	Creates a basic configuration template that supports all the configurations available on the voice register template.
auto-reg-ephone	Enables automatic registration of ephones with the Cisco Unified CME system.

password-persistent

To configure password-persistent option for a vpn-profile, use the **password-persistent** command in vpn-profile configuration mode.

password-persistent [{*enabledisable*}]

Syntax Description

enable	Enables password-persistent to authenticate.
disable	Disables password-persistent to authenticate.

Command Default

Password-persistent is disabled.

Command Modes

Vpn-profile configuration (conf-vpn-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to enable or disable password-persistent option for a vpn-profile.

Examples

The following example shows the password-persistent command enabled for vpn-profile 2:

```
Router#show run
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  keepalive 50
  host-id-check disable
 vpn-profile 2
  mtu 1300
  password-persistent enable
  host-id-check enable
 sip
!
voice class media 10
 media flow-around
!
```

Related Commands

Command	Description
vpn-profile	Defines a VPN-profile.

pattern (voice register dialplan)

To define a dial pattern for a SIP dial plan, use the **pattern** command in voice register dialplan configuration mode. To remove the pattern, use the **no** form of this command.

pattern *tag* *string* [**button** *button-number*] [**timeout** *seconds*] [**user** {**ip** | **phone**}]
no **pattern** *tag*

Syntax Description

tag	Number that identifies the dial pattern. Range: 1 to 24.
string	Dial pattern, such as the area code, prefix, and first one or two digits of the telephone number, plus wildcard characters or dots (.) for the remainder of the dialed digits.
button *button-number*	(Optional) Button to which the dial pattern applies.
timeout *seconds*	(Optional) Time, in seconds, that the system waits before dialing the number entered by the user. Range: 0 to 30. To have the number dialed immediately, specify 0. If this parameter is not used, the phone's default interdigit timeout value is used (10 seconds).
user	(Optional) Tag that automatically gets added to the dialed number. Do not use this keyword if Cisco Unified CME is the only SIP call agent.
ip	(Optional) Sets the value of the user tag to IP in the dialed number.
phone	(Optional) Sets the value of the user tag to phone in the dialed number.

Command Default

No pattern is defined.

Command Modes

Voice register dialplan configuration (config-register-dialplan)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command defines a pattern of dialed digits that are matched by the phone and passed to Cisco Unified CME to initiate a call. Dial strings that match the pattern trigger the sending of a SIP INVITE message to Cisco Unified CME. Patterns are matched sequentially in order of the *tag* number.

You must first use the **type** command to specify the type of phone that the dial plan is being defined for before you can enter a pattern. Enter this command for each dial pattern that is part of the dial plan definition. After you define a dial plan, assign it to a SIP phone by using the **dialplan** command.

The **button** keyword specifies the button to which the dial pattern applies. If the user is initiating a call on line button 1, only the dial patterns specified for button 1 apply. If this keyword is not configured, the dial pattern applies to all lines on the phone. This keyword is not supported on Cisco Unified IP Phones 7905 or 7912. The button number corresponds to the order of the buttons on the side of the screen, from top to bottom, with 1 being the top button.

The **pattern** command and **filename** command are mutually exclusive. You can use either the **pattern** command to define dial patterns manually for a dial plan, or the **filename** command to select a custom dial pattern file that is loaded in system flash.

Examples

The following example shows the dial patterns set for SIP dial plan 10:

```
Router(config)# voice register dialplan 10
Router(config-register-dialplan)# type 7905-7912
Router(config-register-dialplan)# pattern 52...
Router(config-register-dialplan)# pattern 91.......
```

Related Commands

	Description
dialplan	Assigns a dial plan to a SIP phone.
filename	Specifies a custom configuration file that contains dial patterns to use for the SIP dial plan.
show voice register dialplan	Displays all configuration information for a specific SIP dial plan.
type (voice register dialplan)	Defines a phone type for a SIP dial plan.

pattern direct

To configure the dual tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system when a user presses the Messages button on an IP phone in a Cisco CallManager Express (Cisco CME) system, use the **pattern direct** command in voice-mail integration configuration mode. To disable DTMF pattern forwarding when a user presses the Messages button on a phone, use the **no** form of this command.

pattern direct *tag1* {**CDN** | **CGN** | **FDN**} [*tag2* {**CDN** | **CGN** | **FDN**}] [*tag3* {**CDN** | **CGN** | **FDN**}] [*last-tag*]
no pattern direct

Syntax Description

tag1	Alphanumeric string of fewer than four DTMF digits in length. The alphanumeric string can consist of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file and immediately precede the number of the calling party, the number of the called party, or a forwarding number.
CDN	Called number (CDN) information is sent to the voice-mail system.
CGN	Calling number (CGN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.
tag2, *tag3*	(Optional) Same as *tag1*. The router supports a maximum of four tags.
last-tag	(Optional) Same as *tag1*. This tag indicates the end of the pattern.

Command Default

This feature is disabled.

Command Modes

Voice-mail integration configuration (config-vm-integration)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	2.0	This command was introduced
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

The **pattern direct** command is used to configure the sequence of dual tone multifrequency (DTMF) digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is placed directly from a Cisco IP phone attached to the router, the voice-mail system expects to receive a sequence of DTMF digits at the beginning of the call to identify the user's mailbox, accompanied by a string of digits to indicate that the caller is attempting to access the designated mailbox in order to retrieve messages.

Although it is unlikely that you will use multiple instances of the **CDN**, **CGN**, or **FDN** keywords in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for a calling number (CGN) for a direct call to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern direct 2 CGN *
```

Related Commands

	Description
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to a busy extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to an extension that does not answer and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.

pattern ext-to-ext busy

To configure the dual tone multifrequency (DTMF) digit pattern forwarding necessary to activate a voice-mail system after an internal extension attempts to connect to a busy extension and the call is forwarded to voice mail, use the **pattern ext-to-ext busy** command in voice-mail integration configuration mode. To disable the feature, use the **no** form of this command.

pattern ext-to-ext busy *tag1* {**CDN** | **CGN** | **FDN**} [*tag2* {**CDN** | **CGN** | **FDN**}] [*tag3* {**CDN** | **CGN** | **FDN**}] [*last-tag*]
no pattern ext-to-ext busy

Syntax Description

tag1	Alphanumeric string of fewer than four DTMF digits in length. The alphanumeric string can consist of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file and immediately precede the number of the calling party, the number of the called party, or a forwarding number.
CDN	Called number (CDN) information is sent to the voice-mail system.
CGN	Calling number (CGN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.
tag2, *tag3*	(Optional) Same as *tag1*. The router supports a maximum of four tags.
last-tag	(Optional) Same as *tag1*. This tag indicates the end of the pattern.

Command Default

This feature is disabled.

Command Modes

Voice-mail integration configuration (config-vm-integration)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(13)T	Cisco SRST 2.02	This command was added for Cisco SRST.

Usage Guidelines

The **pattern ext-to-ext busy** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is routed to the voice-mail system by call forward on busy from a Cisco IP phone attached to the router, the voice-mail system expects to receive digits that identify the mailbox associated with the forwarding phone together with digits that identify the extension number of the calling IP phone.

Although it is unlikely that you will use multiple instances of the **CDN**, **CGN**, or **FDN** keywords in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for a local call forwarded on busy to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern ext-to-ext busy 7 FDN * CGN *
```

Related Commands

	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the Messages button on the phone.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension that does not answer and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.

pattern ext-to-ext no-answer

To configure the dual tone multifrequency (DTMF) pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to a non answering extension and the call is forwarded to voice mail, use the **pattern ext-to-ext no-answer** command in voice-mail integration configuration mode. To disable this feature, use the **no** form of this command.

pattern ext-to-ext no-answer *tag1* {**CDN** | **CGN** | **FDN**} [*tag2* {**CDN** | **CGN** | **FDN**}] [*tag3* {**CDN** | **CGN** | **FDN**}] [*last-tag*]
no pattern ext-to-ext no-answer

Syntax Description

tag1	Alphanumeric string of fewer than four DTMF digits in length. The alphanumeric string can consist of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file and immediately precede the number of the calling party, the number of the called party, or a forwarding number.
CDN	Called number (CDN) information is sent to the voice-mail system.
CGN	Calling number (CGN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.
tag2, *tag3*	(Optional) Same as *tag1*. The router supports a maximum of four tags.
last-tag	(Optional) Same as *tag1*. This tag indicates the end of the pattern.

Command Default

This feature is disabled.

Command Modes

Voice-mail integration configuration (config-vm-integration)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(13)T	Cisco SRST 2.02	This command was added for Cisco SRST.

Usage Guidelines

The **pattern ext-to-ext no-answer** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is routed to the voice-mail system by call forward on no-answer from an IP phone attached to the router, the voice-mail system expects to receive digits that identify the mailbox associated with the forwarding phone together with digits that identify the extension number of the calling IP phone.

Although it is unlikely that you will use multiple instances of the **CDN**, **CGN**, or **FDN** keywords in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for a local call forwarded on no-answer to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern ext-to-ext no-answer 5 FDN * CGN *
```

Related Commands

	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the Messages button on the phone.
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.

pattern trunk-to-ext busy

To configure the dual tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail, use the **pattern trunk-to-ext busy** command in voice-mail integration configuration mode. To return to the default, use the **no** form of this command.

pattern trunk-to-ext busy *tag1* {**CDN** | **CGN** | **FDN**} [*tag2* {**CDN** | **CGN** | **FDN**}] [*tag3* {**CDN** | **CGN** | **FDN**}] [*last-tag*]
no pattern trunk-to-ext busy

Syntax Description

tag1	Alphanumeric string of fewer than four DTMF digits in length. The alphanumeric string can consist of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file and immediately precede the number of the calling party, the number of the called party, or a forwarding number.
CDN	Called number (CDN) information is sent to the voice-mail system.
CGN	Calling number (CGN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.
tag2, *tag3*	(Optional) Same as *tag1*. The router supports a maximum of four tags.
last-tag	(Optional) Same as *tag1*. This tag indicates the end of the pattern.

Command Default

This feature is disabled by default.

Command Modes

Voice-mail integration configuration (config-vm-integration)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco SRST 2.02	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(13)T	Cisco ITS 2.0	This command was added for Cisco SRST.

Usage Guidelines

The **pattern trunk-to-ext busy** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is routed to the voice-mail system by call forward on busy from an IP phone attached to the router, the voice-mail system expects to receive a sequence of digits identifying the mailbox associated with the forwarding phone together with digits indicating that the call originated from a PSTN or VoIP caller.

Although it is unlikely that you will use multiple instances of the **CDN**, **CGN**, or **FDN** keywords in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for call forwarding when an external trunk call reaches a busy extension and the call is forwarded to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern trunk-to-ext busy 6 FDN * CGN *
```

Related Commands

	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the Messages button on the phone.
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.

pattern trunk-to-ext no-answer

To configure the dual tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail, use the **pattern trunk-to-ext no-answer** command in voice-mail integration configuration mode. To disable this feature, use the **no** form of this command.

pattern trunk-to-ext no-answer *tag1* {**CDN** | **CGN** | **FDN**} [*tag2* {**CDN** | **CGN** | **FDN**}] [*tag3* {**CDN** | **CGN** | **FDN**}] [*last-tag*]
no pattern trunk-to-ext no-answer

Syntax Description

tag1	Alphanumeric string of fewer than four DTMF digits in length. The alphanumeric string can consist of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file and immediately precede the number of the calling party, the number of the called party, or a forwarding number.
CDN	Called number (CDN) information is sent to the voice-mail system.
CGN	Calling number (CGN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.
tag2, *tag3*	(Optional) Same as *tag1*. The router supports a maximum of four tags.
last-tag	(Optional) Same as *tag1*. This tag indicates the end of the pattern.

Command Default

This feature is disabled.

Command Modes

Voice-mail integration configuration (config-vm-integration)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(13)T	2.02	This command was added for Cisco SRST.

Usage Guidelines

The **pattern trunk-to-ext no-answer** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the router through one or more voice ports. When a call is routed to the voice-mail system by call forward on no-answer from an IP phone attached to the router, the voice-mail system expects to receive digits that identify the mailbox associated with the forwarding phone together with digits that indicate that the call originated from a PSTN or VoIP caller.

Although it is unlikely that you will use multiple instances of the **CDN**, **CGN**, or **FDN** keywords in a single command line, it is permissible to do so.

Examples

The following example sets the DTMF pattern for call forwarding when an external trunk call reaches an unanswered extension and the call is forwarded to a voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern trunk-to-ext no-answer 4 FDN * CGN *
```

Related Commands

	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the Messages button on the phone.
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.

phone-display

To enable a phone user to display voice hunt group information using the Services button on the phone, use the **phone-display** command in voice hunt group configuration mode. To remove the configuration, use the **no** form of this command.

phone-display
no phone-display

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this command is disabled.

Command Modes

ephone configuration mode

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

This command when configured, enables the user to view the information of a specific voice hunt group on the phone.

Example

The following example shows how the voice hunt group display option is enabled for a phone :

```
Router(config)# voice hunt-group 1 parallel
Router(config-voice-hunt-group)# phone-display
```

phone-mode only

To enable Jabber phone-only client support, use the **phone-mode only** command. To remove t the configuration, use the **no** form of this command.

phone-mode *phone only*
nophone-mode*phone only*

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this feature is disabled.

Command Modes

voice register global (config-register global)

voice register pool (config-register pool)

voice register template (config-register template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

This command enables Jabber phone-only client support.

Example

The following example shows how phone-mode is enabled:

```
Router(config)# voice register pool
Router(config-register pool)# phone-mode phone-only
```

Related Commands

Command	Description
voice register global	Enters voice register global configuration mode.
voice register pool	Enters voice register pool configuration mode.
voice register template	Enters voice register template configuration mode.

phone-key-size

To specify the size of the RSA key pair that is generated on phones, use the **phone-key-size** command in CAPF-server configuration mode. To return the size to the default, use the **no** form of this command.

phone-key-size {**512** | **1024** | **2048**}
no phone-key-size

Syntax Description

512	512 bits
1024	1024 bits. This is the default key size.
2048	2048 bits

Command Default

RSA key pair size is 1024.

Command Modes

CAPF-server configuration (config-capf-server)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

If you choose a higher key size than the default setting, the phones take longer to generate the entropy that is required to generate the keys. Key generation, which is set at low priority, allows the phone to function while the action occurs. Depending on the phone model, you may notice that key generation takes up to 30 or more minutes to complete.

Examples

The following example specifies a key size of 2048 bits.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

phoneload

To define the phone firmware support for a phone type, use the **phoneload** command in ephone-type configuration mode. To remove firmware support, use the **no** form of this command.

phoneload
no phoneload

Syntax Description

This command has no arguments or keywords.

Command Default

Phone type supports firmware configuration.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies whether the phone type defined in the phone-type template supports firmware configuration using the **load** command.

Examples

The following example shows that support for phone firmware is disabled for the Nokia E61 phone type:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# no phoneload
```

Related Commands

Command	Description
device-name	Assigns a name to a phone type in an ephone-type template.
load	Associates a type of Cisco Unified IP phone with a phone firmware file.

phoneload-support

To define the phone support for firmware download from CME, use the **phoneload-support** command in voice register pool-type mode. To disable phoneload support, use the **no** form of this command.

phoneload-support
noponeload-support

Syntax Description

This command has no arguments or keywords.

Command Default

The phoneload support is disabled. When the **reference-pooltype** command is configured, phoneload support property of the reference phone is inherited.

Command Modes

Voice Register Pool Configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to define the default transport type. If the new phone supports the phoneload, you can use the **load** command in voice register global" mode to configure the corresponding phoneload for the new phone model. When you use the no form of this command, the inherited properties of the reference phone takes precedence over the default value.

Example

The following example shows how to define the phoneload support for a new phone model using the **phoneload-support** command:

```
Router(config)# voice register global
Router(config--register-global)# mode cme
Router(config-register-global)# load 9900 P0S3-06-0-00
Router(config-register-global)# phoneload-support
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.
load	Associates a type of IP phone with a phone firmware file.

phone-redirect-limit (voice register global)

To set the number of 3XX responses an originating SIP phone in a Cisco CallManager Express (Cisco CME) system can accept for a single call, use the **phone-redirect-limit** command in voice register global configuration mode. To revert to the default, use the **no** form of this command.

phone-redirect-limit *number*
no phone-redirect-limit

Syntax Description

number	Maximum number of 3XX responses accepted for a single call. Range: 5 to 20. Default: 5.

Command Default

Default is 5

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

Use this command to control how many subsequent 3XX responses an originating SIP phone can handle for a single call. The terminating side is any forwarding party which does not use B2BUA, but sends 3XX directly to the originating calling phone. When Cisco CME gets a 3XX from the terminating side, Cisco CME relays the 3XX to the originating SIP phone. The default number of 3XXs that the originating phone can accept is 5.

The following example shows how to set the maximum number of redirects to 6:

```
Router(config)# voice register global
Router(config-register-global)# phone-redirect-limit 6
```

Related Commands

	Description
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

phone-ui park-list

To enable a phone user to view the list of active parked calls, use the **phone-ui park-list** command in the ephone configuration mode. To remove the configuration, use the **no** form of this command.

phone-ui park-list
no phone-ui park-list

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this feature is enabled for Skinny Call Control Protocol (SCCP) phones.

Command Modes

ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

This command enables the park-list menu option under My-Phone-Apps service button menu.

Example

The following example shows how to enable the park list display option for phone 7:

```
Router(config)# ephone 7
Router(config-ephone-type)# phone-ui park-list
```

Example

The following example shows how to disable park list display option for phone 7:

```
Router(config)# ephone 7
Router(config-ephone-type)# no phone-ui park-list
```

Related Commands

Command	Description
url button	Enables the configuration of the URL Services feature button on a line key.
url services	Associates a URL with the Programmable Services feature button on the supported Cisco Unified SCCP phones.

phone-ui speeddial-fastdial

To enable a phone user to configure speed-dial and fast-dial numbers from their phone, use the **phone-ui speeddial-fastdial** command in ephone configuration mode. To reset to the default value, use the **no** form of this command.

phone-ui speeddial-fastdial
no phone-ui speeddial-fastdial

Syntax Description

This command has no arguments or keywords.

Command Default

Enabled (speed-dial and fast-dial numbers are configurable from phone).

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command enables the speed-dial and fast-dial configuration menu on the phone so that users can configure these options directly.

The services URL must be configured using the **url services** command.

Examples

The following example shows that the speed-dial and fast-dial user interface is disabled for phone 7:

```
Router(config)# ephone 7
Router(config-ephone-type)# no phone-ui speeddial-fastdial
```

Related Commands

Command	Description
fastdial	Creates an entry for a personal speed-dial number.
speed-dial	Creates speed-dial definitions for a phone.
url services	Associates a URL with the programmable Services feature button on supported Cisco Unified IP phones.

phone-ui voice-hunt-groups

To enable a Skinny Call Control Protocol (SCCP) phone user to display voice hunt group information using the Services button on a phone, use the **phone-ui voice-hunt-groups** command in ephone configuration mode. To remove the configuration, use the **no** form of this command.

phone-ui voice-hunt-groups
no phone-ui voice-hunt-groups

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this command is enabled.

Command Modes

ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

This command enables the Voice Hunt Groups menu option under the My-Phone-Apps service button menu .

Example

The following example shows how to disable the voice hunt group display option for phone 7:

```
Router(config)# ephone 7
Router(config-ephone-type)# no phone-ui voice-hunt-groups
```

Related Commands

Command	Description
url services	Associates a URL with the Programmable Services feature button on supported Cisco Unified IP phones.

pickup-call any-group

To enable a phone user to pickup a ringing call on extensions in any pickup group, use the **pickup-call any-group** command in ephone-dn or voice register dn configuration mode. To reset to the default value, use the **no** form of this command.

pickup-call any-group
no pickup-call any-group

Syntax Description

This command has no arguments or keywords.

Command Default

User can pickup calls in other groups by pressing GPickUp soft key and dialing pickup group number.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command allows a phone user to pickup any ringing call within the local Cisco Unified CME system by pressing the GPickUp soft key and asterisk (*), if the ringing extension is configured with a pickup group using the **pickup-group** command.

If this command is not configured, a phone user can pickup calls only from their local group by pressing the GPickUp soft key and *. To pickup calls in another group, the user must press the GPickUp soft key and dial the pickup group number.

Examples

The following example shows that extension 1020 can pick up calls ringing on extension 1030 by pressing the GPickUp softkey and *:

```
ephone-dn  102
 number 1020
 pickup-call any-group
!
ephone-dn  103
 number 1030
 pickup-group 5
```

Related Commands

Command	Description
pickup-group	Assigns an extension to a call-pickup group.
service directed-pickup	Enables Directed Call Pickup and modifies the function of the GPickUp and PickUp soft keys.
softkeys idle	Modifies the soft-key display on IP phones during the idle call state.

pickup-group

To assign an extension to a call-pickup group, use the **pickup-group** command in ephone-dn, ephone-dn-template, or voice register dn configuration mode. To remove the extension from a group, use the **no** form of this command.

pickup-group *group-number*
no pickup-group

Syntax Description

group-number	String representing a pickup group. The string can contain up to 32 characters.

Command Default

An extension does not belong to any pickup group.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)
Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was added to ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.
12.4(22)YB	Cisco Unified CME 7.1	This command was added to voice register dn configuration mode for SIP directory numbers.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command allows you to assign an individual directory number to a call-pickup group. Phone users can pick up ringing calls within their own pickup group more easily than calls outside their group.

You can assign each directory number to only one pickup group. There is no limit to the number of directory numbers that can be assigned to a single pickup group, and there is no limit to the number of pickup groups that can be defined in a Cisco Unified CME system.

Pickup group numbers can vary in length, but must have unique leading digits. For example, you cannot define pickup group 17 and pickup group 177 in the same Cisco Unified CME system because a pickup in group 17 will always be triggered before the user can enter the final 7 for group 177. You can, however, define pickup groups 27 and 177 in the same Cisco Unified CME system.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following examples assign extension 3242 to pickup group 25:

```
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 3242
Router(config-ephone-dn)# pickup-group 25

Router(config)#  voice register dn 4
Router(config-register-dn)# number 3242
Router(config-register-dn)# pickup-group 25
```

The following example uses an ephone-dn-template to assign extension 3242 to pickup group 25:

```
Router(config)# ephone-dn-template 8
Router(config-ephone-dn-template)# pickup-group 25
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 3242
Router(config-ephone-dn)# ephone-dn-template 8
```

Related Commands

Command	Description
ephone-dn-template (ephone-dn)	Applies a template to an ephone-dn configuration.
service directed-pickup	Enables Directed Call Pickup and modifies the function of the PickUp and GPickUp soft keys.

pilot

To define the ephone-dn that callers dial to reach a Cisco CallManager Express (Cisco CME) ephone hunt group, use the **pilot** command in ephone-hunt configuration mode. To remove the pilot number from the ephone hunt group, use the **no** form of this command.

pilot *number* [**secondary** *number*]
no pilot *number* [**secondary** *number*]

Syntax Description

number	String of up to 27 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters if the number is dialed only by the router, as with an intercom number, or is not intended to be dialed at all. Secondary numbers can contain wildcards in the string. For details, see "Usage Guidelines."
secondary	(Optional) Defines the number that follows as an additional pilot number for the ephone hunt group.

Command Default

No pilot number is defined.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **secondary** *secondary-number* keyword-argument pair was introduced.

Usage Guidelines

This command defines a valid number for an ephone-dn (extension) that is to be assigned to an ephone hunt pilot group. The dial-plan pattern can be applied to the pilot number.

The **secondary** keyword allows you to associate a second telephone number with this ephone-dn so that the hunt group can be called by dialing either the main or secondary phone number. The secondary number may contain one or more wildcards instead of digits, even if the wildcard number overlaps the primary number. For example, 50.. (the number 50 followed by periods, which stand for wildcards) matches all four-digit extensions that start with 50. Wildcard characters cannot be used in the primary pilot number.

Alphabetic characters can be used to create a primary or secondary pilot number that cannot be dialed from a phone and is not part of the dial plan.

Examples

The following example sets the pilot number to 2345 for peer ephone hunt group number 5:

```
ephone-hunt 5 peer
 pilot 2345
 list 2346, 2347, 2348
 hops 3
 timeout 45
```

```
 final 6000
```

The following example sets the pilot number for ephone hunt group 3 to 2222 and the secondary pilot number to 4444:

```
ephone-hunt 3 sequential
 pilot 2222 secondary 4444
 list 2555, 2556, 2557
 final 6000
```

The following example uses wildcards in the secondary pilot number to create a hunt group that receives the calls made to all numbers that start with 555. The primary pilot number, A0, cannot be dialed.

```
ephone-hunt 1 longest-idle
 pilot A0 secondary 555....
 list 1000, 1001, 1002
 timeout 5
 hops 3
 final 1100
```

Related Commands

	Description
ephone-hunt	Enters ephone-hunt configuration mode to define a Cisco CME ephone hunt group.
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Lists the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of an ephone hunt group should not register with the H.323 gatekeeper.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the ephone-hunt-group list.

pilot (voice hunt-group)

To define the number that callers dial to reach a Cisco Unified CME voice hunt group, use the **pilot** command in voice hunt-group configuration mode. To remove the pilot number from the voice hunt group, use the **no** form of this command.

pilot *number* [**secondary** *number*]
no pilot

Syntax Description

number	String of up to 32 characters that represents an extension or E.164 telephone number.
secondary	(Optional) Defines an additional pilot number for the voice hunt group.

Command Default

No pilot number is defined.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command defines an extension that is assigned as the pilot number of a voice hunt group. The dial-plan pattern can be applied to the pilot number.

Normally the pilot number is composed of digits, but the string may contain alphabetic characters if the number is dialed only by the router, as with an intercom number, or is not intended to be dialed at all.

The **secondary** keyword allows you to associate a second telephone number so that the hunt group can be called by dialing either the primary or secondary phone number. The secondary number can contain one or more wild cards instead of digits, even if the wildcard number overlaps the primary number. For example, 50.. (the number 50 followed by periods, which stand for wild card) matches all four-digit extensions that start with 50. Wildcard characters cannot be used in the primary pilot number.

Alphabetic characters can be used to create a primary or secondary pilot number that cannot be dialed from a phone and is not part of the dial plan.

Voice hunt groups do not support the expansion of pilot numbers using the **dialplan-pattern** command. To enable external phones to dial the pilot number, you must configure a secondary pilot number using a fully qualified E.164 number.

Examples

The following example shows how to set the pilot number to 2345 for voice hunt group hunt group number 5:

```
voice-hunt 5 peer
 pilot 2345
 list 2346, 2347, 2348
 hops 3
 timeout 45
 final 6000
```

The following example shows how to set the pilot number for voice hunt group 3 to 2222 and the secondary pilot number to 4444:

```
voice hunt-group 3 sequential
 pilot 2222 secondary 4444
 final 6000
```

The following example shows how to use wild cards in the secondary pilot number to create a voice hunt group that receives the calls made to all numbers that start with 55501. The primary pilot number, A0, cannot be dialed.

```
voice hunt-group 1 longest-idle
 pilot A0 secondary 55501..
 list 1000, 1001, 1002
 timeout 5
 hops 3
 final 1100
```

The following example shows how to use a secondary pilot number in a parallel hunt group. Local phones can dial the primary pilot number, 1100. External phones (PSTN) must dial the full E.164 number, 4085550100.

```
voice hunt-group 4 parallel
 final 1109
 list 1101,1102,1103,1104
 timeout 60
 pilot 1100 4085550100
```

Related Commands

	Description
dialplan-pattern	Defines a pattern that is used to expand extension numbers into fully qualified E.164 numbers.
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next directory number in a peer voice hunt-group list before proceeding to the final directory number.
list (voice hunt-group)	Defines the directory numbers that participate in a hunt group.
voice hunt-group	Defines the type of hunt group.

pin

To set a personal identification number (PIN) for an IP phone in a Cisco CallManager Express (Cisco CME) system, use the **pin** command in ephone configuration mode. To remove a PIN, use the **no** form of this command.

pin *number*
no pin

Syntax Description

number	PIN that will be used to log in to a Cisco IP phone. This is a numeric string from four to eight digits in length.

Command Default

No PIN is set.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The **pin** command allows individual phone users to override call-blocking patterns that are associated with defined time periods. Call-blocking patterns that are in effect at all times (7 days a week, 24 hours a day) cannot be overridden using a PIN.

Call blocking on IP phones is defined in the following way. First, one or more patterns of outgoing digits to be blocked are defined using the **after-hours block pattern** command. Next, one or more time periods during which calls to those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both. By default, all IP phones in a Cisco CME system are restricted if at least one pattern and at least one time period are defined. Individual phones can be completely exempted from call blocking using the **after-hour exempt** command**.** An individual with a PIN can override call blocking by entering the PIN after pressing the Login soft key to log in to a phone that has been configured for that PIN using the **pin** command.

The PIN functionality applies only to IP phones that have soft keys, such as the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G.

Examples

The following example sets a PIN for an IP phone:

```
Router(config)# ephone 1
Router(config-ephone)# pin 1000
```

Related Commands

	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined for a Cisco CME system.

	Description
after-hours block pattern	Defines a pattern of digits to be blocked for outgoing calls from IP phones.
after-hours date	Defines a recurring period based on month and day during which outgoing calls that match defined call-block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined call-block patterns are blocked on IP phones.
login	Defines when IP phones in a Cisco CME system are logged out automatically.
show ephone login	Displays the login states of all phones.

pin (voice logout-profile and voice user-profile)

To configure a personal identification number (PIN) for accessing a particular IP phone that is enabled for extension mobility, use the **pin** command in voice logout-profile configuration mode or voice user-profile configuration mode. To remove a PIN, use the **no** form of this command.

pin **[0|6]** *number*
no **pin** **[0|6]** *number*

Syntax Description

number	Four- to eight-digit numeric string for accessing Cisco Unified IP phone.
[0|6]	The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Command Default

No PIN is configured.

Command Modes

Voice logout-profile configuration (config-logout-profile)
Voice user-profile configuration (config-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(‘1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption.

Usage Guidelines

Use this command in voice logout-profile configuration mode to create a PIN to be used by a phone user to disable the call blocking configuration for a Cisco Unified IP phone on which a logout profile is downloaded.

Use this command in voice user-profile configuration mode to create a PIN to be used by a phone user to disable the call blocking configuration for a Cisco Unified IP phone on which a user profile is downloaded.

PIN functionality applies only to IP phones that have soft keys, such as the Cisco Unified IP Phone 7940 and 7940G.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]** for this command. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Examples

The following example shows the configuration for a user profile to be downloaded when a phone user logs into a Cisco Unified IP phone that is enabled for extension mobility, including a PIN of 12345:

```
pin 12345
user me password pass123
number 2001 type silent-ring
```

```
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

Command	Description
logout-profile	Enable an SCCP phone for Extension Mobility and apply logout profile to phone being configured.
reset (voice logout-profile and voice user-profile)	Performs complete reboot of all IP phones on which a particular logout-profile or user-profile is downloaded.

pin (voice register pool)

To set a personal identification number (PIN) to bypass the after-hour call block on a Cisco Unified SIP IP phone, use the **pin** command in voice register pool configuration mode. To remove the PIN, use the **no** form of the command.

pin **[0|6]** *digits*
no **pin**

Syntax Description

digits	PIN to bypass the after-hour call block on the Cisco Unified SIP IP phone. Numeric string from four to eight digits in length. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Command Default

No valid PIN is set.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
15.2(2)T	Unified CME 9.0	This command was introduced.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption.

Usage Guidelines

The **pin** command allows individual Cisco Unified SIP IP phone users to override call-blocking patterns that are associated with defined time periods. Call-blocking patterns that are in effect at all times (7 days a week, 24 hours a day) cannot be overridden using a PIN.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]** for this command. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Examples

The following example shows how to set a PIN to bypass the after-hour call block on a Cisco Unified SIP IP phone in voice register pool 80:

```
Router(config)# voice register pool 80
Router(config-register-pool)# pin 12345
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for Cisco Unified SIP IP phones in Cisco Unified CME.

port (CAPF-server)

To define the TCP port number on which the CAPF server listens for incoming socket connections, use the **port** command in CAPF-server configuration mode. To use the default, use the **no** form of this command.

port *tcp-port*
no port

Syntax Description

tcp-port	Port for secure communication. Range is from 2000 to 9999. Default is 3804.

Command Default

TCP port number 3804.

Command Modes

CAPF-server configuration (config-capf-server)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example specifies TCP port 3000 instead of the default port 3804:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

preemption reserve timer

To set the amount of time to reserve a channel for a preemption call, use the **preemption reserve timer** command in voice MLPP configuration mode. To reset to the default, use the **no** form of this command.

preemption reserve timer *seconds*
no preemption reserve timer

Syntax Description

seconds	Number of seconds to reserve the channel. Range: 3 to 30. Default: 0.

Command Default

Preemption reserve timer is disabled (0).

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

When a channel on a SCCP phone is preempted by a higher priority MLPP call, the channel is reserved for the MLPP call so that other calls cannot use that channel before the call is connected.

Examples

The following example shows the reserve timer set to 10 seconds.

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# preemption reserve timer 10
```

Related Commands

Command	Description
preemption enable	Enables preemption capabilities on a trunk group.
preemption tone timer	Sets the expiry time for the preemption tone for the outgoing call being preempted by a DDR backup call.
preemption user	Enables phones to preempt calls.

preemption tone timer (voice MLPP)

To set the amount of time the preemption tone plays on the called phone when a lower precedence call is being preempted, use the **preemption tone timer** command in voice MLPP configuration mode. To reset to the default, use the **no** form of this command.

preemption tone timer *seconds*
no preemption tone timer

Syntax Description

seconds	Length of preemption tone, in seconds. Range: 3 to 30. Default: 0.

Command Default

Preemption tone timer is disabled (0).

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

This command sets how long a phone user hears the preemption tone play when a lower precedence call is being preempted by a higher priority call. The preemption tone stops playing when the timer expires or the user goes onhook.

For calls to Cisco Unified IP phones, the called party can hang up immediately to connect to the new higher precedence call, or if the called party does not hang up, Cisco Unified CME forces the phone on-hook after the preemption tone timer expires and connects the call.

For FXS ports, the called party must acknowledge the preemption by going on-hook, before being connected to the new higher precedence call.

The **mlpp indication** command must be enabled (default) for a phone to play preemption tones.

Examples

The following example shows the tone timer is set to 15 seconds:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# preemption tone timer 15
```

Related Commands

Command	Description
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
mlpp preeemption	Enables the preemption capability on an SCCP phone or analog FXS port.
preemption reserve timer	Sets the amount time to reserve a channel for a preemption call.
preemption user	Enables the preemption capability for all supported phones.

preemption trunkgroup

To enable preemption capabilities for trunk groups, use the **preemption trunkgroup** command in voice MLPP configuration mode. To disable preemption capabilities, use the **no** form of this command.

preemption trunkgroup
no preemption trunkgroup

Syntax Description

This command has no arguments or keywords.

Command Default

Preemption is disabled for trunk groups.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

The following example enables preemption capabilities for trunk groups:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# preemption trunkgroup
```

Related Commands

Command	Description
mlpp preemption	Enables calls on an SCCP phone or analog FXS port to be preempted.
preemption user	Enables phones to preempt calls.

preemption user

To enable phones to preempt calls, use the **preemption user** command in voice MLPP configuration mode. To disable preemption capabilities, use the **no** form of this command.

preemption user
no preemption user

Syntax Description

This command has no arguments or keywords.

Command Default

Preemption is disabled for phones.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Products	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

This command enables SCCP and analog FXS phones in the system to preempt calls if the called party is busy with lower precedence calls.

Examples

The following example enables preemption capabilities for phones:

```
Router(config)# voice mlpp
Router(conf-voi-mlpp)# preemption user
```

Related Commands

Command	Description
mlpp preemption	Enables preemption capabilities on an SCCP phone or analog FXS port.
preemption trunkgroup	Enables preemption capabilities on a trunk group.

preference (ephone-dn)

To set dial-peer preference order for an extension (ephone-dn) associated with a Cisco IP phone, use the **preference** command in ephone-dn configuration mode. To reset the preference order to the default, use the **no** form of this command.

preference *preference-order* [**secondary** *secondary-order*]
no preference

Syntax Description

preference-order	Preference order for the primary number associated with an extension (ephone-dn). Type **?** for a range, where 0 is the highest preference. Default is 0.
secondary *secondary-order*	(Optional) Preference order for the secondary number associated with the ephone-dn. Type **?** for a range, where 0 is the highest preference. Default is 9.

Command Default

Preference order for the primary number is 0 (highest preference). Preference order for the secondary number is 9 (lowest preference).

Command Modes

Ephone-dn configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)ZJ	Cisco CME 3.0	The **secondary** *secondary-order* keyword-argument pair was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

When you create an ephone-dn for an IP phone in a Cisco CallManager Express (Cisco CME) system, you automatically create a virtual voice port and one to four virtual dial peers to be used by that ephone-dn. This command sets a preference value for the primary and secondary numbers that are associated with the ephone-dn that you are creating. The preference values are passed transparently into the dial peer or dial peers created by the ephone-dn. The preference values allow you to control the selection of a desired dial peer when multiple dial peers are matched on the same destination-pattern (target) number value. In this way, the **preference** command can be used to establish a hunt strategy for incoming calls.

The **huntstop** command can be used to prevent further hunting for a dial-peer match when an ephone-dn is busy or does not answer.

Examples

The following example sets a preference of 2 for the directory number 3000:

```
ephone-dn 1
 number 3000
 preference 2
```

In the following example, the number 1222 under ephone-dn 4 has a higher preference than the number 1222 under ephone-dn 5.

```
ephone-dn 4
 number 1222
 preference 0
!
!
ephone-dn 5
 number 1222
 preference 1
```

The following example shows an ephone-dn with two numbers. The primary number has a higher preference than the secondary number.

```
ephone-dn 6
 number 2233 secondary 2234
 preference 0 secondary 1
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
huntstop	Discontinues call hunting behavior for an extension (ephone-dn) or an extension channel.

preference (ephone-hunt)

To set preference order for the ephone-dn associated with an ephone-hunt-group pilot number in Cisco Unified CME, use the **preference** command in ephone-hunt configuration mode. To delete this preference order, use the **no** form of this command.

preference *preference-order* [**secondary** *secondary-order*]
no preference *preference-order* [**secondary** *secondary-order*]

Syntax Description

preference-order	Preference order. Range is 0 to 8, where 0 is the highest preference and 8 is the lowest preference. Default is 0.
secondary *secondary-order*	(Optional) Preference order for the secondary pilot number. Range is 1 to 8, where 1 is the highest preference and 8 is the lowest preference. Default is 7.

Command Default

Preference order for the primary number is 0 (highest preference). Preference order for the secondary number is 7 (lowest preference).

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **secondary** *secondary-order* keyword-argument pair was introduced.

Usage Guidelines

This command sets a preference value that is used for matching dial peers in a Cisco IP phone virtual dial-peer group. The preference value is associated with a pilot number for a Cisco CME ephone hunt group. The preference value is passed transparently into the dial peer created by the pilot number. Setting the preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

Examples

The following example sets the preference for the pilot number of hunt group 23 to 1:

```
Router(config)# ephone-hunt 23 sequential
Router(config-ephone-hunt)# pilot 2355
Router(config-ephone-hunt)# preference 1
```

Related Commands

Command	Description
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.

Command	Description
list	Lists the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in an Cisco CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of an ephone hunt group not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that callers dial to reach an ephone hunt group.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the ephone-hunt-group list.

preference (voice hunt-group)

To set preference order for the voice dial peer associated with a voice hunt-group pilot number in Cisco Unified CME, use the **preference** command in voice hunt-group configuration mode. To delete this preference order, use the **no** form of this command.

preference *preference-order* [**secondary** *secondary-order*]
no preference *preference-order* [**secondary** *secondary-order*]

Syntax Description

preference-order	Preference order for the extension or telephone number associated with a dial peer. Range is 0 to 8. Default is 0.
secondary *secondary-order*	(Optional) Preference order for the secondary pilot number. Range is 1 to 8, where 1 is the highest preference and 8 is the lowest preference. Default is 7.

Command Default

Preference for primary number is 0 (highest preference). Preference for secondary number is 7 (lower preference).

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command sets a preference value that is used for matching dial peers in a Cisco IP phone virtual dial-peer group. The preference value is associated with a pilot number for a Cisco CME voice hunt group. The preference value is passed transparently into the dial peer created by the pilot number. Setting the preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

Note It is recommended that the parallel hunt-group pilot number be unique in the system. Parallel hunt groups may not work if there are more than one partial or exact dial-peer match. For example, this happens if the pilot number is “8000” and there is another dial peer that matches “8...”. If multiple matches cannot be avoided, give call parallel hunt group the highest priority to run by assigning a lower preference to the other dial peers. Note that 8 is the lowest preference value. By default, dial peers created by parallel hunt groups have a preference of 0.

Examples

The following is an example of a parallel voice hunt group. The pilot number is 6000 and the preference assigned to the pilot number is 1:

```
voice hunt-group 2 parallel
 pilot 6000
 preference 1
 list 3000, 3010, 3020
 final 9999
 timeout 10
```

Related Commands

Command	Description
pilot (voice hunt-group)	Defines the voice dn that callers dial to reach a Cisco CallManager Express (Cisco CME) voice hunt group.
voice hunt-group	Defines the type of hunt group.

preference (voice register dn)

To set the dial-peer preference order for VoIP dial peer to be created for a directory number on a SIP phone, use the **preference** command in voice register dn configuration mode. To reset the preference order to the default, use the **no** form of this command.

preference *preference-order*
no preference

Syntax Description

preference-order	Preference order for the extension or telephone number associated with a directory number. Range is 0 to 10. Default is 0.

Command Default

Preference for primary number is 0 (highest preference).

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

When you create a directory number for a SIP phone in a Cisco CallManager Express (Cisco CME) or Cisco SIP SRST environment, you automatically create a virtual voice port and one to four virtual dial peers to be used by that directory number. This command sets a preference value for the extension or telephone number that is associated with the directory number hat you are creating. The preference value is passed transparently to dial peers created by the directory number. The preference value allows you to control the selection of a desired dial peer when multiple dial peers are matched on the same destination pattern (extension or telephone number). In this way, the **preference** command can be used to establish a hunt strategy for incoming calls.

The **huntstop** command can be used to prevent further hunting for a dial-peer match when a number is busy or does not answer.

Note This command can also be used for Cisco SIP SRST.

Examples

The following example shows how to set a preference of 2 for extension number 3000:

```
voice register dn 1
 number 3000
 preference 2
```

In the following example, extension number 1222 under voice register dn 4 has a higher preference than number 1222 under voice register dn 5.

```
voice register dn 4
 number 1222
 preference 0
!
```

```
!
voice register dn 5
 number 1222
 preference 1
```

Related Commands

	Description
huntstop (voice register dn)	Discontinues call hunting behavior for an extension (directory number) or an extension channel.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

preference (voice register pool)

To set the preference order for creating the VoIP dial peers created for a number associated with a voice pool, use the **preference** command in voice register pool configuration mode. To put the number in default preference order, use the **no** form of this command.

preference *preference-order*
no preference

Syntax Description

preference-order	Preference order for the extension or telephone number associated with a pool. Range is 0 to 10. Default is 0, which is the highest preference.

Command Default

Preference for primary number is 0 (highest preference order).

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CallManager Express (Cisco CME).

Usage Guidelines

When you create a voice register pool for a SIP phone or a group of SIP phones in a Cisco Unified CME or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment, you automatically create a virtual voice port and one to four virtual dial peers to be used by the number associated with that pool. The preference value is passed transparently to dial peers created for the number. The preference value allows you to control the selection of a desired dial peer when multiple dial peers are matched on the same destination pattern (extension or phone number) associated with the pool. In this way, the **preference** command can be used to establish a hunt strategy for incoming calls.

Note Configure the **id** (voice register pool) command before any other voice register pool commands, including the preference command. The id command identifies a locally available individual SIP phone or set of Cisco SIP phones.

Examples

The following example shows how to set a preference of 2 for extension number 3000:

```
voice register pool 1
 number 3000
 preference 2
```

In the following example, extension number 1222 under voice register dn 4 has a higher preference than number 1222 under voice register pool 5.

```
voice register pool 4
 number 1222
 preference 0
!
!
voice register dn 5
 number 1222
 preference 1
```

Related Commands

	Description
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
voice register pool	Enters voice register pool configuration mode for SIP phones.

presence

To enable presence service and enter presence configuration mode, use the **presence** command in global configuration mode. To disable presence service, use the **no** form of this command.

presence
no presence

Syntax Description

This command has no arguments or keywords.

Command Default

Presence service is disabled.

Command Modes

Global configuration (config)

Command History

Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command enables the router to perform the following presence functions:

- Process presence requests from internal lines to internal lines. Notify internal subscribers of any status change.
- Process incoming presence requests from a SIP trunk for internal lines. Notify external subscribers of any status change.
- Send presence requests to external presentities on behalf of internal lines. Relay status responses to internal lines.

Examples

The following example shows how to enable presence and enter presence configuration mode to set the maximum subscriptions to 150:

```
Router(config)# presence
Router(config-presence)# max-subscription 150
```

Related Commands

	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
debug presence	Displays debugging information about the presence service.
max-subscription	Sets the maximum number of concurrent watch sessions that are allowed.
presence enable	Allows the router to accept incoming presence requests.
server	Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.

	Description
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.

presence call-list

To enable Busy Lamp Field (BLF) monitoring for call lists and directories on phones registered to the Cisco Unified CME router, use the **presence call-list** command in ephone, presence, or voice register pool configuration mode. To disable BLF indicators for call lists, use the **no** form of this command.

presence call-list
no presence call-list

Syntax Description

This command has no arguments or keywords.

Command Default

BLF monitoring for call lists is disabled.

Command Modes

Ephone configuration (config-ephone)
Presence configuration (config-presence)
Voice register pool configuration (config-register pool)

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command enables a phone to monitor the line status of directory numbers listed in a directory or call list, such as a missed calls, placed calls, or received calls list. Using this command in presence mode enables the BLF call-list feature for all phones. To enable the feature for an individual SCCP phone, use this command in ephone configuration mode. To enable the feature for an individual SIP phone, use this command in voice register pool configuration mode.

If this command is disabled globally and enabled in voice register pool or ephone configuration mode, the feature is enabled for that voice register pool or ephone.

If this command is enabled globally, the feature is enabled for all voice register pools and ephones regardless of whether it is enabled or disabled on a specific voice register pool or ephone.

To display a BLF status indicator, the directory number associated with a telephone number or extension must have presence enabled with the **allow watch** command.

For information on the BLF status indicators that display on specific types of phones, see the Cisco Unified IP Phone documentation for your phone model.

Examples

The following example shows the BLF call-list feature enabled for ephone 1. The line status of a directory number that appears in a call list or directory is displayed on phone 1 if the directory number has presence enabled.

```
Router(config)# ephone 1
Router(config-ephone)# presence call-list
```

Related Commands

	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
blf-speed-dial	Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.
presence	Enables presence service and enters presence configuration mode.
show presence global	Displays configuration information about the presence service.

presence enable

To allow incoming presence requests, use the **presence enable** command in SIP user-agent configuration mode. To block incoming requests, use the **no** form of this command.

presence enable
no presence enable

Syntax Description

This command has no arguments or keywords.

Command Default

Incoming presence requests are blocked.

Command Modes

SIP UA configuration (config-sip-ua)

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command allows the router to accept incoming presence requests (SUBSCRIBE messages) from internal watchers and SIP trunks. It does not impact outgoing presence requests.

Examples

The following example shows how to allow incoming presence requests:

```
Router(config)# sip-ua
Router(config-sip-ua)# presence enable
```

Related Commands

	Description
allow subscribe	Allows internal watchers to monitor external presence entities (directory numbers).
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
max-subscription	Sets the maximum number of concurrent watch sessions that are allowed.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.
watcher all	Allows external watchers to monitor internal presence entities (directory numbers).

present-call

To present ephone-hunt-group calls only to member phones that are idle or onhook, use the **present-call** command in ephone-hunt configuration mode. To return to the default, use the **no** form of this command.

present-call {**idle-phone** | **onhook-phone**}
no present-call {**idle-phone** | **onhook-phone**}

Syntax Description

idle-phone	Presents calls from the ephone-hunt group only if all lines are idle on the phone on which the hunt-group line appears. This option does not consider monitored lines that have been configured on the phone using the **button m** command.
onhook-phone	Presents calls from the ephone-hunt group only if the phone on which the number appears is in the onhook state. When this keyword is configured, calls in the ringing or hold state that are unrelated to the hunt group do not prevent the presentation of calls from the ephone-hunt group.

Command Default

Ephone hunt group calls are presented to lines (ephone-dns) that are not in use, regardless of the state of other lines on the same ephone.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

If you do not use this command, an ephone hunt group presents calls to an ephone whenever the phone line (ephone-dn) that corresponds to a number in an ephone-hunt list is available. The status of other phone lines on the phone is not considered.

The **present-call** command adds additional controls that allow you to take into account the activity on all lines of a phone that has an ephone-dn that is assigned to an ephone hunt group. The **present-call** command allows you to specify that hunt groups should present calls to these phones only when they are on hook or are not busy with an active call. This keeps hunt group calls from possibly going unanswered because a phone is occupied with a call on a line other than the line assigned to the hunt group.

Examples

The following example sets up a peer hunt group with three ephone-dns to answer calls. Incoming calls are sent only to ephone-dns on phones that are on-hook.

```
ephone-hunt 17 peer
pilot 3000
list 3011, 3021, 3031
hops 3
final 7600
present-call onhook-phone
```

Related Commands

	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.

present-call (voice hunt-group)

To present voice hunt-group calls only to member phones that are idle, use the **present-call** command in voice hunt group configuration mode. To return to the default, use the **no** form of this command.

present-call {**idle-phone**}
no present-call {**idle-phone**}

Syntax Description

idle-phone	Presents calls from the voice hunt group only if all lines are idle on the phone on which the hunt group line appears.

Command Default

Voice hunt group calls are presented to lines (ephone-dns or voice register dns) that are not in use, regardless of the state of other lines on the same ephone or voice register pool.

Command Modes

voice hunt-group configuration (config-voice-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1 15.6(3)M1	Cisco Unified CME 11.6	This command was introduced for voice hunt group.

Usage Guidelines

If you do not use this command, voice hunt group presents calls to an ephone or voice register pool whenever the phone line (ephone-dn or voice register dn) that corresponds to a number in a voice hunt group list is available. The status of other phone lines on the phone is not considered.

The **present-call** command adds additional controls that allow you to take into account the activity on all lines of a phone that has an ephone-dn or voice register dn that is assigned to a voice hunt group. The **present-call** command allows you to specify that hunt groups should present calls to these phones only when they are idle or not busy with an active call. This keeps hunt group calls from possibly going unanswered because a phone is occupied with a call on a line other than the line assigned to the hunt group.

Examples

The following example sets up a peer hunt group with three ephone-dns to answer calls. Incoming calls are sent only to ephone-dns or voice register dns on phones that are idle.

```
voice hunt-group 17 peer
pilot 3000
list 3011, 3021, 3031
final 7600
present-call idle-phone
```

Related Commands

	Description
voice hunt-group	Defines a voice hunt group and enters voice hunt-group configuration mode.

privacy (ephone)

To modify privacy support on a specific phone, use the **privacy** command in ephone or ephone-template configuration mode. To reset to the default value, use the **no** form of this command.

privacy [{**off** | **on**}]
no privacy

Syntax Description

off	(Optional) Disables privacy on the phone.
on	(Optional) Enables privacy on the phone.

Command Default

Use system-level setting configured with the **privacy** command in telephony-service mode.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command modifies the privacy capability of individual phones. Privacy prevents other phone users from seeing call information or barging into a call on a shared octo-line directory number. Privacy is supported for calls on shared octo-line directory numbers only.

If only specific phones require access to privacy, disable privacy at the system-level by using the **no privacy** command in telephony-service configuration mode and enable privacy at the phone-level by using the **privacy on** command.

After a phone that is configured for privacy registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. If the button has a lamp, it lights. When the phone receives an incoming call, the user can make the call private by pressing the Privacy feature button. The privacy button toggles between on and off. The privacy state is applied to new calls and current calls that the user owns.

Users can dynamically enable privacy for shared-line calls by pressing the Privacy feature button on the phone if the **privacy-button** command is enabled.

The Privacy feature applies to all shared lines on a phone. If a phone has multiple shared lines and Privacy is enabled, other phones cannot view or barge into calls on any of the shared lines.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode for the same ephone, the value that you set in ephone configuration mode has priority.

Examples

The following example shows privacy enabled on a specific phone and disabled at the system-level:

```
telephony-service
 no privacy
```

```
 privacy-on-hold
 max-ephones 100
 max-dn 240
!
!
ephone  10
 privacy on
 privacy-button
 max-calls-per-button 3
 busy-trigger-per-button 2
 mac-address 00E1.CB13.0395
 type 7960
 button  1:7 2:10
```

Related Commands

Command	Description
privacy (telephony-service)	Enables privacy globally for all phones in the system.
privacy-button	Enables the privacy feature button on an IP phone.
privacy-on-hold	Enables privacy for calls that are on hold on shared octo-line directory numbers.

privacy (telephony-service)

To enable privacy at the system level for all phones, use the **privacy** command in telephony-service configuration mode. To disable privacy at the system level, use the **no** form of this command.

privacy
no privacy

Syntax Description

This command has no arguments or keywords.

Command Default

Privacy is enabled at the system level for all phones.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command enables privacy for all phones in the system. Privacy prevents other phone users from seeing call information or joining a call on a shared octo-line directory number. Privacy is supported for calls on shared octo-line directory numbers only.

If only specific phones need access to privacy, disable privacy at the system-level by using the **no privacy** command and enable privacy at the phone level by using the **privacy on** command in ephone or ephone-template configuration mode.

After a phone that is configured for privacy registers with Cisco Unified CME, the feature button on the phone is labeled "Privacy" and a status icon displays. The button lamp, if available, lights to reflect the privacy setting of the phone. When the phone receives an incoming call, the user can make the call private by pressing the Privacy feature button.

Examples

The following example shows privacy disabled at the system-level and enabled on an individual phone:

```
telephony-service
 no privacy
 privacy-on-hold
 max-ephones 100
 max-dn 240
 timeouts transfer-recall 60
 voicemail 8900
 max-conferences 8 gain -6
 transfer-system full-consult
 fac standard
!
!
ephone  10
 privacy on
 privacy-button
 max-calls-per-button 3
```

```
busy-trigger-per-button 2
mac-address 00E1.CB13.0395
type 7960
button  1:7 2:10
```

Related Commands

Command	Description
privacy (ephone)	Modifies privacy support on a specific phone.
privacy-button	Enables the privacy feature button on an IP phone.
privacy-on-hold	Enables privacy for calls that are on hold on shared octo-line directory numbers.

privacy (voice register global)

To enable privacy at the system level for all SIP phones, use the **privacy** command in voice register global configuration mode. To disable privacy at the system level, use the **no** form of this command.

privacy
no privacy

Syntax Description

This command has no arguments or keywords.

Command Default

Privacy is enabled at the system level for all phones.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command enables privacy for all phones in the system. Privacy prevents other phone users from seeing call information or joining a call on a shared-line directory number. Privacy is supported for calls on shared-line directory numbers only.

If only specific phones need access to privacy, disable privacy at the system-level by using the **no privacy** command and enable privacy at the phone level by using the **privacy on** command in voice register pool or voice register template configuration mode.

After a phone that is configured with the **privacy-button** command registers with Cisco Unified CME, the feature button on the phone is labeled "Privacy" and a status icon displays. The button lamp, if available, lights to reflect the privacy setting of the phone. When the phone receives an incoming call, the user can make the call private by pressing the Privacy feature button.

Examples

The following example shows privacy disabled at the system-level and enabled on an individual phone:

```
voice register global
 mode cme
 privacy-on-hold
 no privacy
 max-dn 300
 max-pool 150
 voicemail 8900
 call-feature-uri pickup http://10.4.212.11/pickup
 call-feature-uri gpickup http://10.4.212.11/gpickup
!
voice register pool  130
 id mac 001A.A11B.500E
 type 7941
 number 1 dn 30
 template 6
```

```
dnd
privacy ON
```

Related Commands

Command	Description
privacy (voice register pool)	Modifies privacy support on a specific phone.
privacy-button	Enables the privacy feature button on an IP phone.
privacy-on-hold	Enables privacy for calls that are on hold on shared-line directory numbers.
shared-line	Creates a shared-line directory number for a SIP phone.

privacy (voice register pool)

To modify the phone-level privacy setting on a SIP phone, use the **privacy** command in voice register pool or voice register template configuration mode. To reset to the default value, use the **no** form of this command.

privacy {**off** | **on**}
no privacy

Syntax Description

off	Disables privacy on the phone.
on	Enables privacy on the phone.

Command Default

Use system-level setting configured with the **privacy** command in voice register global mode.

Command Modes

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command modifies the privacy setting on the SIP phone. Privacy prevents other phone users from viewing call information or barging into a call on a shared-line directory number. Privacy is supported for calls on shared-line directory numbers only.

After a phone that is configured with the **privacy-button** command registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. If the button has a lamp, it lights. When the phone receives an incoming call, the user can make the call private by pressing the Privacy feature button. The privacy button toggles between on and off. The privacy state applies to new calls and current calls that the phone user owns.

The **off** and **on** keywords specify the initial Privacy state on the phone when the Privacy feature is enabled. The phone user can then toggle the privacy state on and off using the Privacy feature button.

The Privacy state applies to all shared lines on a phone. If a phone has multiple shared lines, other phones cannot view or barge into calls on any of the shared lines if the Privacy state is enabled.

If you use a voice register template to apply a command to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

Examples

The following example shows privacy enabled for a specific SIP phone:

```
Router(config)# voice register pool 123
Router(config-register-pool)# privacy on
```

Related Commands

Command	Description
privacy (voice register global)	Enables privacy at the system level for all SIP phones.
privacy-button	Enables the privacy feature button on an IP phone.
privacy-on-hold	Enables privacy for calls that are on hold on shared-line directory numbers.
shared-line	Creates a shared-line directory number for a SIP phone.
softkeys remote-in-use (voice register template)	Modifies the soft-key display during the remote-in-use call state on SIP phones.
template (voice register pool)	Applies a template to a SIP phone.

privacy-button

To enable the privacy feature button on an IP phone, use the **privacy-button** command in ephone, ephone-template, voice logout-profile, and voice user-profile configuration mode. To reset to the default value, use the **no** form of this command.

privacy-button
no privacy-button

Syntax Description

This command has no arguments or keywords.

Command Default

Privacy button is disabled.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Voice logout-profile configuration (config-logout-profile)
Voice user-profile configuration (config-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command allows phone users to dynamically enable or disable privacy for calls on shared octo-lines by pressing the Privacy feature button on the phone. Privacy prevents other phone users from viewing call information or joining calls on a shared octo-line directory number.

Privacy is supported only for calls on shared octo-line directory numbers so enable this command only on phones that share an octo-line directory number.

After a phone that is configured for privacy registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. The button lamp, if available, lights to reflect the privacy setting of the phone. The privacy feature button toggles between on and off. The privacy state is applied to new calls and current calls owned by the user.

Privacy is enabled on the phone with either the **privacy** command in ephone configuration mode or the **privacy** command in telephony-service mode.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode for the same ephone, the value that you set in ephone configuration mode has priority.

Examples

The following example shows the privacy button is enabled for ephone 10:

```
ephone  10
 privacy-button
 max-calls-per-button 3
 busy-trigger-per-button 2
 mac-address 00E1.CB13.0395
```

```
type 7960
button  1:7
```

Related Commands

Command	Description
privacy (ephone)	Modifies privacy support on a specific phone.
privacy (telephony-service)	Enables privacy globally for all phones in the system.
privacy-on-hold	Enables privacy for calls that are on hold on shared octo-line directory numbers.

privacy-button (voice register pool)

To enable the privacy feature button on an IP phone, use the **privacy-button** command in voice register pool or voice register template configuration mode. To reset to the default value, use the **no** form of this command.

privacy-button
no privacy-button

Syntax Description

This command has no arguments or keywords.

Command Default

Privacy button is disabled.

Command Modes

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command allows phone users to dynamically enable or disable privacy for calls on shared lines by pressing the Privacy feature button on the phone. Privacy prevents other phone users from viewing call information or joining calls on a shared-line directory number.

Privacy is supported only for calls on shared-line directory numbers so enable this command only on phones that use a shared-line directory number.

After a phone that is configured with this command registers with Cisco Unified CME, the feature button on the phone is labeled “Privacy” and a status icon displays. The button lamp, if available, lights to reflect the privacy setting of the phone. The privacy feature button toggles between on and off. The privacy state is applied to new calls and current calls owned by the user.

Privacy is enabled on the phone with either the **privacy** command in voice register pool configuration mode or the **privacy** command in voice register global configuration mode.

If you use a voice register template to apply a command to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

Examples

The following example shows the privacy button is enabled for phone 124:

```
voice register pool  124
 busy-trigger-per-button 5
 id mac 0017.E033.0284
 type 7965
 number 1 dn 24
 privacy-button
```

Related Commands

Command	Description
privacy (voice register global)	Enables privacy globally for all SIP phones in the system.
privacy (voice register pool)	Modifies privacy support on a specific phone.
privacy-on-hold (voice register global)	Enables privacy for calls that are on hold on shared-line directory numbers.
shared-line	Creates a shared-line directory number for a SIP phone.

privacy-on-hold

To enable privacy for calls that are on hold on shared octo-line directory numbers, use the **privacy-on-hold** command in telephony-service configuration mode. To disable privacy for calls on hold, use the **no** form of this command.

privacy-on-hold
no privacy-on-hold

Syntax Description

This command has no arguments or keywords.

Command Default

Privacy on hold is disabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command prevents other phone users from seeing or retrieving calls that are on hold on a shared octo-line directory number.

Privacy is enabled on the phone with either the **privacy** command in ephone configuration mode or the **privacy** command in telephony-service mode.

Examples

The following example shows how to enable privacy on hold for shared lines.

```
Router(config)# telephony-service
Router(config-telephony)# privacy-on-hold
```

Related Commands

Command	Description
privacy (ephone)	Modifies privacy support on a specific phone.
privacy (telephony-service)	Enables privacy globally for all phones in the system.
privacy-button	Enables the privacy feature button on an IP phone.

privacy-on-hold (voice register global)

To enable privacy for calls that are on hold on shared-line directory numbers, use the **privacy-on-hold** command in voice register global configuration mode. To disable privacy for calls on hold, use the **no** form of this command.

privacy-on-hold
no privacy-on-hold

Syntax Description

This command has no arguments or keywords.

Command Default

Privacy on hold is disabled.

Command Modes

Voice register global (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.

Usage Guidelines

This command prevents other phone users from seeing or retrieving calls that are on hold on a shared-line directory number.

Privacy is enabled on the phone with either the **privacy** command in voice register pool configuration mode or the **privacy** command in voice register global configuration mode.

Examples

The following example shows how to enable privacy on hold for shared lines.

```
Router(config)# voice register global
Router(config-register-global)# privacy-on-hold
```

Related Commands

Command	Description
privacy (voice register global)	Enables privacy globally for all phones in the system.
privacy (voice register pool)	Modifies privacy support on a specific phone.
privacy-button (voice register pool)	Enables the privacy feature button on an IP phone.
shared-line	Creates a shared-line directory number for a SIP phone.

protocol mode

To configure the Cisco IOS Session Initiation Protocol (SIP) stack, use the **protocol mode** command in SIP user-agent configuration mode. To disable the configuration, use the **no** form of this command.

protocol mode {**ipv4** | **ipv6** | **dual-stack** [**preference** {**ipv4** | **ipv6**}]}
no protocol mode

Syntax Description

ipv4	Specifies the IPv4-only mode.
ipv6	Specifies the IPv6-only mode.
dual-stack	Specifies the dual-stack (that is, IPv4 and IPv6) mode.
preference {**ipv4** | **ipv6**	(Optional) Specifies the preferred dual-stack mode, which can be either IPv4 (the default preferred dual-stack mode) or IPv6.

Command Default

No protocol mode is configured. The Cisco IOS SIP stack operates in IPv4 mode when the **no protocol mode** or **protocol mode ipv4** command is configured.

Command Modes

SIP user-agent configuration (config-sip-ua)

Command History

Release	Modification
12.4(22)T	This command was introduced.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

The **protocol mode** command is used to configure the Cisco IOS SIP stack in IPv4-only, IPv6-only, or dual-stack mode. For dual-stack mode, the user can (optionally) configure the preferred family, IPv4 or IPv6.

For a particular mode (for example, IPv6-only), the user can configure any address (for example, both IPv4 and IPv6 addresses) and the system will not hide or restrict any commands on the router. SIP chooses the right address for communication based on the configured mode on a per-call basis.

For example, if the domain name system (DNS) reply has both IPv4 and IPv6 addresses and the configured mode is IPv6-only (or IPv4-only), the system discards all IPv4 (or IPv6) addresses and tries the IPv6 (or IPv4) addresses in the order they were received in the DNS reply. If the configured mode is dual-stack, the system first tries the addresses of the preferred family in the order they were received in the DNS reply. If all of the addresses fail, the system tries addresses of the other family.

Examples

The following example configures dual-stack as the protocol mode:

```
Router(config-sip-ua)# protocol mode dual-stack
```

The following example configures IPv6 only as the protocol mode:

```
Router(config-sip-ua)# protocol mode ipv6
```

The following example configures IPv4 only as the protocol mode:

```
Router(config-sip-ua)# protocol mode ipv4
```

The following example configures no protocol mode:

```
Router(config-sip-ua)# no protocol mode
```

Related Commands

Command	Description
sip ua	Enters SIP user-agent configuration mode.

protocol-mode (telephony-service)

To configure a preferred IP address mode for SCCP IP phones in Cisco Unified CMEr, use rthe **protocol mode** command in telephony service configuration mode. To disable the router protocol mode, use the **no** form of this command.

protocol mode {**ipv4** | **ipv6** | **dual-stack** [**preference** {**ipv4** | **ipv6**}]}
no protocol mode {**ipv4** | **ipv6** | **dual-stack** [**preference** {**ipv4** | **ipv6**}]}

Syntax Description

ipv4	IPv4-only mode.
ipv6	IPv6-only mode.
dual-stack	Dual-stack mode, ipv4 and ipv6 mode.
preference	Preference dual-stack mode, either ipv4 or ipv6 mode.

Command Default

No protocol mode is configured

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.

Usage Guidelines

The **protocol mode** command is used to configure SCCP IP phones in CUCME in IPv4-only, IPv6-only, or dual-stack mode. For dual-stack mode, the user can configure the preferred family, IPv4 or IPv6.

For a specific mode, the user is free to configure any address and the system will not hide or restrict any commands on the router. On a per-call basis, SCCP phones choose the right address for communication based on the configured mode.

For example, if the DNS reply has both IPv4 and IPv6 addresses and the configured mode is IPv6-only (or IPv4-only), the system discards all IPv4 (or IPv6) addresses and tries the IPv6 (or IPv4) addresses in the order they were received in the DNS reply. If the configured mode is dual-stack, the system first tries the addresses of the preferred family in the order they were received in the DNS reply. If all of the addresses fail, the system tries addresses of the other family.

Examples

The following example configures dual-stack as the protocol mode:

```
Router(config)# telephony-service
Router(config-telephony)#protocol mode dual-stack preference ?
  ipv4  IPv4 address is prefered
  ipv6  IPv6 address is prefered
```

The following example configures IPv6-only mode as the protocol mode:

```
Router(config)# telephony-service
Router(config-telephony)#protocol mode ipv6
```

The following example configures IPv4-only mode as the protocol mode:

```
Router(config)# telephony-service
Router(config-telephony)#protocol mode ipv6
```

Related Commands

Command	Description
ip source-address	Identifies the IP address and port through which IP phones communicate with a Cisco Unified CME router.
shutdown	Allows to shut down SCCP server listening sockets.

provision-tag

To create a provision tag for identifying an ephone or voice register pool for the extension assigner application, use the **provision-tag** command in ephone configuration mode or voice register pool configuration mode. To remove the provision tag, use the **no** form of this command.

provision-tag *tag*
no provision-tag *tag*

Syntax Description

tag	Unique number that identifies this provision tag. Range: 1 to 2147483647.

Command Default

No provision tag is created.

Command Modes

Ephone configuration (config-ephone)

Voice register Pool (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
Cisco IOS XE Everest 16.4.1 15.6(3)M1	Cisco Unified CME 11.6	This command was supported under voice register pool for SIP phones.

Usage Guidelines

This command creates a provision tag.

For SCCP phones, a provision tag enables you to use some number other than an ephone tag, such as a jack number or an extension number, to identify an ephone configuration. The provision tag can be used with the extension assigner application to assign the corresponding ephone configuration to an IP phone. This command is ignored unless you also use the **extension-assigner tag-type** command with the **provision-tag** keyword.

Examples

The following example shows that provision tag 1001 is configured for ephone 1 and provision tag 1002 is configured for ephone 2:

```
Telephony-service
 extension-assigner tag-type provision-tag
 auto assign 101-102
 auto-reg-ephone
Ephone-dn 101
 number 1001
Ephone-dn 102
 number 1002
Ephone 1
 provision-tag 1001
 mac-address 02EA.EAEA.0001
```

```
 button 1:101
Ephone 2
 provision-tag 1002
 mac-address 02EA.EAEA.0002
 button 1:102
```

Examples

For SIP phones, a provision tag enables you to assign any number within the range as an extension number. The provision tag is used with an extension assigner application to assign the corresponding voice register pool configuration to an IP phone.

The following example shows that provision tag 1001 is configured for voice register pool 1 and provision tag 1002 is configured for voice register pool 2:

```
Voice register global
 auto-register
  password xxxx
  auto assign 101-102
voice register dn 101
 number 1001
voice register dn 102
 number 1002
voice register pool 1
 provision-tag 1001
 mac-address 02EA.EAEA.0001
 number 1 dn 101
voice register pool  2
 provision-tag 1002
 mac-address 02EA.EAEA.0002
 number 2 dn 102
```

Related Commands

	Description
extension-assigner tag-type	Specifies which type of tag is used by the extension assigner application to identify an ephone configuration.

Cisco Unified CME Commands: R

refer target dial-peer

To populate the Refer To portion of a SIP Refer message with the address from the dial peer for the directory number being configured, use the **refer target dial-peer** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

refer target dial-peer
no refer target

Syntax Description

This command has no arguments or keywords.

Command Default

Call is transferred to the destination as specified in the SIP Refer message.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW2	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command in voice register dn configuration mode to specify that the destination address for this directory number be the dial peer. If this command is not configured, Cisco IOS software will transfer the call to the destination in the SIP Refer message and if that destination address is Cisco Unified CME, call SIP will send out and route back to CME before sending to the directory number, creating two extra call legs.

The following partial output from the **show working-configuration** command shows the configuration for three directory numbers. This configuration will populate the Refer To portion of the SIP Refer message with the address from the dial peer for each of the directory numbers.

```
voice register dn  1
 session-server 1
 number 8999
 allow watch
 refer target dial-peer
!
voice register dn  2
 session-server 1
 number 8001
 allow watch
 refer target dial-peer
!
voice register dn  3
 session-server 1
 number 8101
 allow watch
 refer target dial-peer
```

refer-ood enable

To enable out-of-dialog refer (OOD-R) processing, use the **refer-ood enable** command SIP user-agent configuration mode. To disable OOD-R, use the **no** form of this command.

refer-ood enable [*request-limit*]
no refer-ood enable

Syntax Description

request-limit	(Optional) Maximum number of concurrent incoming OOD-R requests that the router can process. Range: 1 to 500. Default: 500.

Command Default

OOD-R processing is disabled.

Command Modes

SIP UA configuration (config-sip-ua)

Command History

Release	Cisco product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Out of dialog Refer allows applications to establish calls using the SIP gateway or Cisco Unified CME. The application sets up the call and the user does not dial out from their own phone.

Examples

The following example shows how to enable OOD-R:

```
Router(config)# sip-ua
Router(config-sip-ua)# refer-ood enable
```

Related Commands

	Description
authenticate (voice register global)	Defines the authenticate mode for SIP phones in a Cisco Unified CME or Cisco Unified SRST system.
credential load	Reloads a credential file into flash memory.
debug voip application	Displays all application debug messages.

reference-pooltype

To inherit the properties from the nearest supported phone model for a Cisco Unified SIP IP phone on Cisco Unified CME, use the **reference-pooltype** command in voice register pooltype mode. To remove the pool-type configuration, use the **no** form of this command.

reference- pooltype phone-type
noreference- pooltype phone-type

Syntax Description

phone-type	Unique number that represents the phone model.

Command Default

There is no reference phone to inherit the properties. This is the only command which has the **no** form as the default form.

Command Modes

Voice Register Pool Configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to inherit the properties from the nearest supported phone model for a Cisco Unified SIP IP phone on Cisco Unified CME.

Example

The following example shows how to enter voice register pool configuration mode and inherit the properties from the nearest supported phone model SIP phones on a Cisco Unified CME system:

```
Router# configure terminal
Router(config)# voice register pool-type 9900
Router(config--register-pool-type)# reference-pooltype 9971
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.

regenerate (ctl-client)

To create a new CTLFile.tlv file after making changes to the CTL client configuration, use the **regenerate** command in CTL-client configuration mode. The **no** form of this command has no effect in the configuration.

regenerate
no regenerate

Syntax Description

This command has no arguments or keywords.

Command Default

A new CTLFile.tlv file is not created until this command is used.

Command Modes

CTL-client configuration (config-ctl-client)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example gives the instruction to regenerate the CTL file with the current information.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

register-id

To create an ID for explicitly identifying an external feature server during Register requests, use the **register-id** command in voice register session-server configuration mode. To remove an ID, use the **no** form of this command.

register-id *name*
no register-id *name*

Syntax Description

name	String of up to 30 alphanumeric characters.

Command Default

No identifier is created.

Command Modes

Voice register session-server configuration (config-register-fs)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW2	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to create an ID for identifying a route point during Register requests. Cisco Unified CME challenges and authenticates the initial keepalive Register request and issues a system-wide unique Cisco-referenceID to be included in the response to the Register request from this route point.

Examples

The following partial output shows the configuration of a session manager for an external feature server, including the register ID of CSR1:

```
router# show running-configuration

!
!
voice register session-server 1
 register-id CSR1
 keepalive 360
```

Related Commands

Command	Description
keepalive	Duration for registration after which the registration expires unless the feature server reregisters before the registration expiry.

registrar server (SIP)

To enable SIP registrar functionality, use the **registrar server** command in SIP configuration mode. To disable SIP registrar functionality, use the **no** form of the command.

registrar server [**expires** [**max** *sec*] [**min** *sec*]]
no registrar server

Syntax Description

expires	(Optional) Sets the active time for an incoming registration.
max *sec*	(Optional) Maximum expires time for a registration, in seconds. The range is from 600 to 86400. The default is 3600.
min *sec*	(Optional) Minimum expires time for a registration, in seconds. The range is from 60 to 3600. The default is 60.

Command Default

SIP registrar functionality on the Cisco Unified CME router id disabled.

Command Modes

SIP configuration (config-voi-sip)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was added to Cisco CME.

Usage Guidelines

When this command is entered, the router accepts incoming SIP Register messages. If SIP Register message requests are for a shorter expiration time than what is set with this command, the SIP Register message expiration time is used.

This command is mandatory for Cisco Unified SIP SRST or Cisco Unified CME and must be entered before any **voice register pool** or **voice register global** commands are configured.

If the WAN is down and you reboot your Cisco Unified CME or Cisco Unified SIP SRST router, when the router reloads it will have no database of SIP phone registrations. The SIP phones will have to register again, which could take several minutes, because SIP phones do not use a keepalive functionality. To shorten the time before the phones re-register, the registration expiry can be adjusted with this command. The default expiry is 3600 seconds; an expiry of 600 seconds is recommended.

Examples

The following partial sample output from the **show running-config** command shows that SIP registrar functionality is set:

```
voice service voip
 allow-connections sip-to-sip
 sip
```

```
registrar server expires max 1200 min 300
```

Related Commands

	Description
sip	Enters SIP configuration mode from voice service VoIP configuration mode.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.
voice register pool	Enters voice register pool configuration mode for SIP phones.

reset (ephone)

To perform a complete reboot of a single phone associated with a Cisco CallManager Express (Cisco CME) router, use the **reset** command in ephone configuration mode.

reset

Syntax Description

This command has no arguments or keywords.

Command Default

No reset is performed.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T

Usage Guidelines

After you update information for one or more phones associated with a Cisco CME router, the phone or phones must be rebooted. There are two commands to reboot the phones: **reset** and **restart.** The **reset** command performs a “hard” reboot similar to a power-off-power-on sequence. It reboots the phone and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server to update from their information as well. The **restart** command performs a “soft” reboot by simply rebooting the phone without contacting the DHCP and TFTP servers. The **reset** command takes significantly longer to process than the **restart** command when you are updating multiple phones, but it must be used after updating phone firmware, user locale, network locale, or URL parameters. For simple button, line, or speed-dial changes, you can use the **restart** command.

Use the **reset** (ephone) command to perform a complete reboot of an IP phone when you are in ephone configuration mode. This command has the same effect as a **reset** (telephony-service) command that is used to reset a single phone.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example resets the SCCP phone with a phone-tag of 1:

```
Router(config)# ephone 1
Router(config-ephone)# reset
```

Related Commands

	Description
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

reset (telephony-service)

To perform a complete reboot of one or all phones associated with a Cisco CallManager Express (Cisco CME) router, use the **reset** command in telephony-service configuration mode. To interrupt and cancel a sequential reset cycle, use the **no** form of the command with the **sequence-all** keyword.

reset {**all** [*time-interval*] | **cancel***mac-address* | **sequence-all**}
no reset {**all** [*time-interval*] | **cancel***mac-address* | **sequence-all**}

Syntax Description

all	Resets all Cisco IP phones served by the Cisco CME router. The router pauses for 15 seconds between the reset starts for each successive phone unless the *time-interval* argument is used to change that value.
time-interval	(Optional) Time interval, in seconds, between each phone reset. Range is from 0 to 60. Default is 15.
cancel	Interrupts a sequential reset cycle that was started with a **reset sequence-all** command.
mac-address	MAC address of a particular Cisco IP phone.
sequence-all	Resets all phones in strict one-at-a-time order by waiting for one phone to reregister before starting the reset for the next phone. The sequencing of resets prevents possible conflicts between phones trying to access TFTP services simultaneously. There is a reset timeout of 4 minutes, after which the router stops waiting for the currently registering phone to complete registration and starts to reset the next phone.

Command Default

No reset is performed.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(11)YT	Cisco ITS 2.1	The *time-interval* range maximum was increased from 15 to 60 and the default was changed from 0 to 15.
12.2(11)YT1	Cisco ITS 2.1	The **cancel** and **sequence-all** keywords were introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

After you update information for one or more phones associated with a Cisco CME router, the phone or phones must be rebooted using either the **reset** command or the **restart** command**.** The **reset** command performs a "hard" reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information as well. The **restart** command performs a "soft" reboot by simply rebooting the phone without contacting the DHCP and TFTP servers. The **reset** command takes significantly longer to process than the **restart** command when you are updating multiple phones, but it must be used after you make changes to phone firmware, user locale, network locale, or URL parameters. For simple button, line, or speed-dial changes, you can use the **restart** command.

When you use the **reset** command, the default time interval of 15 seconds is recommended so that phone reset operations are staggered in order to avoid all phones attempting to access router system resources at the same time. A shorter interval may be used on systems with only a small number of phones or for cases where a simple reset of the phones is desired that does not result in the phones downloading updates to the phone firmware (using the router's TFTP service).

When you use the **reset sequence-all** command, the router waits for one phone to complete its reset and reregister before starting to reset the next phone. The delay provided by this command prevents multiple phones from attempting to access the TFTP server simultaneously and therefore failing to reset properly. Each reset operation can take several minutes when you use this command. There is a reset timeout of 4 minutes, after which the router stops waiting for the currently registering phone to complete registration and starts to reset the next phone.

If the router configuration is changed so that the eXtensible Markup Language (XML) configuration files for the phones are modified (changes are made to user locale, network locale, or phone firmware), then whenever you use the **reset all** or **restart all** command, the router automatically executes the **reset sequence-all** command instead. The **reset sequence-all** command resets phones one at a time in order to prevent multiple phones from trying to contact the TFTP server simultaneously. This one-at-a-time sequencing can take a long time if there are many phones. To avoid this automatic behavior, use the **reset all** *time-interval* or the **restart all** *time-interval* with an explicit argument that is not equal to the default 15-second time interval; for example, set a time interval of 14 seconds. If a **reset sequence-all** command has been started in error, use the **reset cancel** command to interrupt and cancel the sequence of resets.

The **restart** command allows the system to perform quick phone resets in which only the button template, line information, and speed-dial information is updated. See the documentation for the **restart** command for more information.

The **no** form of this command has an effect only when used with the **all** or **sequence-all** keyword, when it interrupts and cancels the sequential resetting of phones.

Examples

The following example resets all IP phones served by the Cisco CME router:

```
Router(config)# telephony-service
Router(config-telephony)# reset all
```

The following example resets the Cisco IP phone with the MAC address CFBA.321B.96FA:

```
Router(config)# telephony-service
Router(config-telephony)# reset CFBA.321B.96FA
```

The following example resets all IP phones in sequential, not-overlapping order:

```
Router(config)# telephony-service
Router(config-telephony)# reset sequence-all
```

Related Commands

	Description
reset (ephone)	Performs a complete reboot of a single phone associated with a Cisco CME router.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.

	Description
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.
telephony-service	Enters telephony-service configuration mode.

reset (voice logout-profile and voice user-profile)

To perform a complete reboot of all IP phones on which a particular extension-mobility profile is downloaded, use the **reset** command in voice logout-profile configuration mode or voice user-profile configuration mode.

reset

Syntax Description

This command has no arguments or keywords.

Command Default

No reset is performed.

Command Modes

Voice logout-profile configuration (voice-logout-profile)
Voice user-profile configuration (voice-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW2	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to perform a "hard" reboot similar to a power-off-power-on sequence, which includes downloading updated information from the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server.

Configure this command in voice logout-profile configuration mode after creating or modifying a logout profile for extension mobility.

Configure this command in voice user-profile configuration mode after creating or modifying an individual user's profile for extension mobility.

This command has a no form, but the no form has no effect.

Examples

The following example shows how to modify a logout profile by adding speed-dial definitions and then reset all IP phones on which this logout profile is downloaded to propagate the modification:

```
Router# configure terminal
Router(config)# voice logout-profile
12
Router(config-user-profile)# speed-dial 1 3001
Router(config-user-profile)# speed-dial 2 3002 blf
Router (config-logout-profile)# reset
Router (config-logout-profile)# exit
Router(config)#
```

reset (voice register global)

To perform a complete reboot of all SIP phones associated with a Cisco CallManager Express (Cisco CME) router, use the **reset** command in voice register global configuration mode.

reset

Syntax Description

This command has no arguments or keywords.

Command Default

No reset is performed.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

After you update information for one or more SIP phones associated with a Cisco CME router, reboot the phones by using the **reset** command. The **reset** command performs a "hard" reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information as well. Configure the **reset** command after you make changes to phone firmware, user locale, network locale, or URL parameters.

The time interval between each phone reset is 15 seconds, thereby avoiding an attempt by all phones to access router system resources at the same time.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example shows how to reset all SIP phones served by the Cisco CME router:

```
Router(config)# voice register global
Router(config-register-global)# reset
```

Related Commands

	Description
reset (voice register pool)	Performs a complete reboot of a single SIP phone associated with a Cisco CME router.

reset (voice register pool)

To perform a complete reboot of a specific SIP phone associated with a Cisco CallManager Express (Cisco CME) router, use the **reset** command in voice register pool configuration mode. To interrupt a reset cycle, use the **no** form of this command.

reset
no reset

Syntax Description

This command has no arguments or keywords.

Command Default

No reset is performed.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

After you update information for one or more phones associated with a Cisco CME router, the phones must be rebooted by using the **reset** command. The **reset** command performs a "hard" reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information as well. Configure the **reset** command after you make changes to phone firmware, user locale, network locale, or URL parameters.

Use this command to perform a complete reboot of an individual SIP phone when you are in voice register pool configuration mode. To reset all SIP phones, use the **reset** (voice register global) command.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example shows how to reset SIP phone 1 served by the Cisco CME router:

```
Router(config)# voice register pool 1
Router(config-register-pool)# reset
```

Related Commands

	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.

reset (voice-gateway)

To perform a complete reboot of all analog phones associated with the voice gateway and registered to Cisco Unified CME, use the **reset** command in voice-gateway configuration mode.

reset

Syntax Description

This command has no arguments or keywords.

Command Default

No reset is performed.

Command Modes

Voice-gateway configuration (config-voice-gateway)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

After you update information for one or more analog phones associated with the voice gateway, reboot the phones by using the **reset** command. The **reset** command performs a "hard" reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information. Use the **reset** command after you make changes to phone firmware, user or network locales, or URL parameters.

The time interval between each phone reset is 15 seconds, to avoid an attempt by all phones to access system resources at the same time.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example shows how to reset all analog phones associated with the voice gateway:

```
Router(config)# voice-gateway system 1
Router(config-voice-gateway)# reset
```

Related Commands

Command	Description
restart (voice-gateway)	Performs a fast restart of all analog endpoints associated with the voice gateway.

reset tapi

To reset the connection between a Telephony Application Programmer's Interface (TAPI) application and a particular SCCP phone in Cisco Unified CME, use the **reset tapi** command in ephone configuration mode.

reset tapi

Syntax Description

This command has no arguments or keywords.

Command Default

No reset of the connection between the TAPI application and the router is performed.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(20)YA	Cisco Unified CME 7.0(1)	This command was introduced.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command in ephone configuration mode resets the connection between a TAPI application and a particular SCCP phone. This command does not reset the Ethernet phone.

To disassociate and reestablish the connection without using this command, you must reboot the router.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example shows how to reset the connection between a TAPI application and the SCCP phone associated with the ephone-tag of 1:

```
Router(config)# ephone 1
Router(config-ephone)# reset tapi
```

restart (ephone)

To perform a fast reboot of an IP phone associated with a Cisco CallManager Express (Cisco CME) router, use the **restart** command in ephone configuration mode. To cancel the reboot, use the **no** form of this command.

restart
no restart

Syntax Description

This command has no arguments or keywords.

Command Default

No restart is performed.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT1	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command causes the system to perform a fast phone reboot in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the **reset** command**.** The **restart** command is much faster than the **reset** command because the phone does not need to access the DHCP or TFTP server.

To restart all phones in a Cisco CME system for quick changes to buttons, lines, and speed-dial numbers, use the **restart** command in telephony-service configuration mode.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example restarts the phone with phone-tag 1:

```
Router(config)# ephone 1
Router(config-ephone)# restart
```

Related Commands

	Description
reset (ephone)	Performs a complete reboot of a Cisco IP phone associated with a Cisco CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

restart (telephony-service)

To perform a fast reboot of one or all phones associated with a Cisco CallManager Express (Cisco CME) router, use the **restart** command in telephony-service configuration mode. To cancel the reboot, use the **no** form of this command.

restart {**all** [*time-interval*]*mac-address*}
no restart {**all** [*time-interval*]*mac-address*}

Syntax Description

all	Restarts all phones associated with the Cisco CME router.
time-interval	(Optional) Time between each phone restart, in seconds. Range is from 0 to 60. Default is 15.
mac-address	MAC address of the phone to be restarted.

Command Default

Time-interval is 15 seconds.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT1	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command causes the system to perform a fast phone reset in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the **reset** command.

Use the **restart** command to reboot IP phones after quick changes to buttons, lines, and speed-dial numbers. This command is much faster than the **reset** command because the phone does not access the DHCP or TFTP server.

To restart a single phone, use the **restart c**ommand with the *mac-address* argument or use the **restart** command in ephone configuration mode.

If the router configuration is changed so that the eXtensible Markup Language (XML) configuration files for the phones are modified (changes are made to user locale, network locale, or phone firmware), then whenever you use the **reset all** or **restart all** command, the router automatically executes the **reset sequence-all** command instead. The **reset sequence-all** command resets phones one at a time in order to prevent multiple phones trying to contact the TFTP server simultaneously. This one-at-a-time sequencing can take a long time if there are many phones. To avoid this automatic behavior, use the **reset all** *time-interval* command or the **restart all** *time-interval* command with an explicit argument that is not equal to the default 15-second time interval; for example, set a time interval of 14 seconds. If a **reset sequence-all** command has been started in error, use the **reset cancel** command to interrupt and cancel the sequence of resets.

The **no** form of this command has an effect only when used with the **all** keyword, when it interrupts and cancels the sequential restarting of phones.

Examples

The following example performs a quick restart of all phones in a Cisco CME system:

```
Router(config)# telephony-service
Router(config-telephony)# restart all
```

Related Commands

	Description
reset (ephone)	Performs a complete reboot of a Cisco IP phone associated with a Cisco CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.

restart (voice register)

To perform a fast reset of one or all SIP phones associated with a Cisco Unified CME router, use the **restart** command in voice register global or voice register pool configuration mode. To cancel the reboot, use the **no** form of this command.

restart
no restart

Syntax Description

This command has no arguments or keywords.

Command Default

SIP phones are not restarted.

Command Modes

Voice register global configuration (config-register-global)
Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command causes the system to perform a fast phone reset in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the **reset** command.

Use this command to reboot SIP phones after quick changes to buttons, lines, and speed-dial numbers. This command is much faster than the **reset** command because the phone does not access the DHCP or TFTP server.

To restart a single SIP phone, use the **restart** command in voice register pool configuration mode. To restart all SIP phones in a Cisco Unified CME system, use the **restart** command in voice register global configuration mode.

This command has a **no** form, however the **no** form has no effect.

Note

This command requires firmware load 8-0-2-14 or later versions; it is not supported in older SIP phone loads. To support this command on SIP phones using older firmware, you must upgrade all your phone firmware.

Examples

The following example performs a quick restart of all SIP phones in a Cisco Unified CME system:

```
Router(config)# voice register global
Router(config-register-global)# restart
```

The following example performs a quick restart of SIP phone 10:

```
Router(config)# voice register pool 10
Router(config-register-pool)# restart
```

Related Commands

	Description
reset (voice register pool)	Performs a complete reboot of a single SIP phone associated with a Cisco Unified CME router.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.

restart (voice-gateway)

To perform a fast restart of all analog phones associated with the voice gateway and registered to Cisco Unified CME, use the **restart** command in voice-gateway configuration mode.

restart

Syntax Description

This command has no arguments or keywords.

Command Default

Analog phones are not restarted.

Command Modes

Voice-gateway configuration (config-voice-gateway)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command initiates a quick phone restart in which only the buttons, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user and network locales, or URL parameters, use the **reset** command.

Use this command to reboot all analog phones on the voice gateway after simple configuration changes to buttons, lines, and speed-dial numbers. This command is faster than the **reset** command because the phone does not access the DHCP server.

This command has a **no** form, although the **no** form has no effect.

Examples

The following example shows how to perform a quick restart of all analog phones:

```
Router(config)# voice-gateway system 1
Router(config-voice-gateway)# restart
```

Related Commands

Command	Description
reset (voice-gateway)	Performs a complete reboot of all analog endpoints associated with the voice gateway.

ring (ephone-dn)

To set the ring pattern for all incoming calls to an ephone-dn, use the **ring** command in ephone-dn configuration mode. To return to the standard ring pattern, use the **no** form of this command.

ring {**external** | **feature** | **internal**} [{**primary** | **secondary**}]
no ring

Syntax Description

external	External ring pattern is used for all incoming calls.
feature	Feature ring pattern is used for all incoming calls.
internal	Internal ring pattern is used for all incoming calls.
primary	(Optional) Ring pattern is used on primary number only.
secondary	(Optional) Ring pattern is used on secondary number only.

Command Default

Standard ring pattern is used.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command allows you to select one of the three ring styles supported by SCCP—internal, external, or feature ring. The ring pattern is used for all types of incoming calls to this directory number, on all phones on which the directory number appears. If the phone is already in use, an incoming call is presented as a call-waiting call and uses the distinctive call-waiting beep.

If the **primary** or **secondary** keyword is used, the distinctive ring is used only if the incoming called number matches the primary number or secondary number defined for the ephone-dn. If there is no secondary number defined for the ephone-dn, the **secondary** keyword has no effect.

By default, Cisco Unified CME uses the internal ring pattern for calls between local IP phones and uses the external ring pattern for all other types of calls.

You can associate the feature ring pattern with a specific button on a phone by using the **button f** command. This command assigns the ring pattern to the button on the phone so that different phones that share the same directory number can use a different ring style.

Examples

The following example sets external ringing for all incoming calls on extension 2389.

```
ephone-dn 24
 number 2389
 ring external
```

Related Commands

	Description
button	Associates ephone-dns with individual buttons on an IP phone and specifies line type or ring behavior.

route-code

To enable phone users to dial a route code to specify special routing for a call, use the **route-code** command in voice MLPP configuration mode. To reset to the default, use the **no** form of this command.

route-code
no route-code

Syntax Description

This command has no arguments or keywords.

Command Default

Route code is disabled.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command enables users to specify special routing for an MLPP call by dialing a route code. The route code is a two-digit number beginning with 1.

Examples

The following example shows how to enable users to dial a route code:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# route-code
```

Related Commands

Command	Description
access-digit	Defines the access digit that phone users dial to request a precedence call.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.
service-digit	Enables phone users to dial a service digit to request off-net services.

rule (voice translation-rule)

To define a translation rule, use the **rule** command in voice translation-rule configuration mode. To delete the translation rule, use the **no** form of this command.

Match and Replace Rule
rule*precedence /match-pattern/ /replace-pattern/***[type** *{match-type replace-type}***[plan***{match-type replace-type}***]]**
no rule *precedence*

Reject Rule
rule*predence***reject***precedence /match-pattern/ /replace-pattern//match-pattern/***[type** *{match-type***[plan***{match-type r***]]**
no rule *precedence*

Syntax Description

precedence	Priority of the translation rule. Range is from 1 to 15.
/match-pattern/	Stream editor (SED) expression used to match incoming call information. The slash '/' is a delimiter in the pattern.
/replace-pattern/	SED expression used to replace the match pattern in the call information. The slash '/' is a delimiter in the pattern.
type *match-type replace-type*	(Optional) Number type of the call. Valid values for the *match-type* argument are as follows: • **abbreviated**—Abbreviated representation of the complete number as supported by this network. • **any**—Any type of called number. • **international**—Number called to reach a subscriber in another country. • **national**—Number called to reach a subscriber in the same country, but outside the local network. • **network**—Administrative or service number specific to the serving network. • **reserved**—Reserved for extension.**subscriber**—Number called to reach a subscriber in the same local network. • **unknown**—Number of a type that is unknown by the network. Valid values for the *replace-type* argument are as follows: • **abbreviated**—Abbreviated representation of the complete number as supported by this network. • **international**—Number called to reach a subscriber in another country. • **national**—Number called to reach a subscriber in the same country, but outside the local network.
type *match-type replace-type* (continued)	• **network**—Administrative or service number specific to the serving network. • **reserved**—Reserved for extension. • **subscriber**—Number called to reach a subscriber in the same local network. • **unknown**—Number of a type that is unknown by the network.

plan *match-type replace-type*	(Optional) Numbering plan of the call. Valid values for the *match-type* argument are as follows: • **any**—Any type of dialed number. • **data** • **ermes** • **isdn** • **national**—Number called to reach a subscriber in the same country, but outside the local network. • **private** • **reserved**—Reserved for extension. • **telex** • **unknown**—Number of a type that is unknown by the network. Valid values for the *replace-type* argument are as follows: • **data** • **ermes** • **isdn** • **national**—Number called to reach a subscriber in the same country, but outside the local network. • **private** • **reserved**—Reserved for extension. • **telex** • **unknown**—Number of a type that is unknown by the network.
reject	The match pattern of a translation rule is used for call-reject purposes.

Command Default

No default behavior or values

Command Modes

Voice translation-rule configuration

Command History

Release	Modification
12.2(11)T	This command was introduced with a new syntax in voice-translation-rule configuration mode.
15.1(4)M	This command was introduced with an increase in the maximum value of the precidence variable from 15 to 100.

Usage Guidelines

Note Use this command in conjunction after the **voice translation-rule** command. An earlier version of this command uses the same name but is used after the **translation-rule** command and has a slightly different command syntax. In the older version, you cannot use the square brackets when you are entering command syntax. They appear in the syntax only to indicate optional parameters, but are not accepted as delimiters in actual command entries. In the newer version, you can use the square brackets as delimiters. Going forward, we recommend that you use this newer version to define rules for call matching. Eventually, the **translation-rule** command will not be supported.

A translation rule applies to a calling party number (automatic number identification [ANI]) or a called party number (dialed number identification service [DNIS]) for incoming, outgoing, and redirected calls within Cisco H.323 voice-enabled gateways.

Number translation occurs several times during the call routing process. In both the originating and terminating gateways, the incoming call is translated before an inbound dial peer is matched, before an outbound dial peer is matched, and before a call request is set up. Your dial plan should account for these translation steps when translation rules are defined.

The below table shows examples of match patterns, input strings, and result strings for the rule (voice translation-rule) command.

Table 13: Match Patterns, Input Strings and Result Strings

Match Pattern	Replacement Pattern	Input String	Result String	Description
/^.*/	//	4085550100		Any string to null string.
//	//	4085550100	4085550100	Match any string but no replacement. Use this to manipulate the call plan or call type.
/\(^...\)456\(...\)/	/\1555\2/	4084560177	4085550177	Match from the middle of the input string.
/\(.*\)0120/	/\10155/	4081110120	4081110155	Match from the end of the input string.
/^1#\(.*\)/	/\1/	1#2345	2345	Replace match string with null string.
/^408...\(8333\)/	/555\1/	4087770100	5550100	Match multiple patterns.
/1234/	/00&00/	5550100	55500010000	Match the substring.
/1234/	/00\000/	5550100	55500010000	Match the substring (same as &).

The software verifies that a replacement pattern is in a valid E.164 format that can include the permitted special characters. If the format is not valid, the expression is treated as an unrecognized command.

The number type and calling plan are optional parameters for matching a call. If either parameter is defined, the call is checked against the match pattern and the selected type or plan value. If the call matches all the conditions, the call is accepted for additional processing, such as number translation.

Several rules may be grouped together into a translation rule, which gives a name to the rule set. A translation rule may contain up to 15 rules. All calls that refer to this translation rule are translated against this set of criteria.

The precedence value of each rule may be used in a different order than that in which they were typed into the set. Each rule's precedence value specifies the priority order in which the rules are to be used. For example, rule 3 may be entered before rule 1, but the software uses rule 1 before rule 3.

The software supports up to 128 translation rules. A translation profile collects and identifies a set of these translation rules for translating called, calling, and redirected numbers. A translation profile is referenced by trunk groups, source IP groups, voice ports, dial peers, and interfaces for handling call translation.

Examples

The following example applies a translation rule. If a called number starts with 5550105 or 70105, translation rule 21 uses the rule command to forward the number to 14085550105 instead.

```
Router(config)# voice translation-rule 21
 Router(cfg-translation-rule)# rule 1 /^5550105/ /14085550105/
 Router(cfg-translation-rule)# rule 2 /^70105/ /14085550105/
```

In the next example, if a called number is either 14085550105 or 014085550105, after the execution of translation rule 345, the forwarding digits are 50105. If the match type is configured and the type is not “unknown,” dial-peer matching is required to match the input string numbering type.

```
Router(config)# voice translation-rule 345
 Router(cfg-translation-rule)# rule 1 /^14085550105/ /50105/ plan any national
 Router(cfg-translation-rule)# rule 2 /^014085550105/ /50105/ plan any national
```

Related Commands

Command	Description
show voice translation-rule	Displays the parameters of a translation rule.
voice translation-rule	Initiates the voice translation-rule definition.

Cisco Unified CME Commands: S1

sast1 trustpoint

To specify the name of the SAST1 trustpoint, use the **sast1 trustpoint** command in CTL-client configuration mode. To return to the default, use the **no** form of this command.

sast1 trustpoint *label*
no sast1

Syntax Description

label	Name of the SAST1 trustpoint.

Command Default

No SAST1 trustpoint name is specified

Command Modes

CTL-client configuration (config-ctl-client)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

The **sast1 trustpoint** and **sast2 trustpoint** commands are used to set up the System Administrator Security Token (SAST) credentials, which are used to sign the CTL file. The SAST1 and SAST2 certificates must be different from each other, but to conserve memory either one of them can use the same certificate as Cisco Unified CME. The CTL file is always signed by SAST1 credentials. The SAST2 credentials are included in the CTL file so that if the SAST1 certificate is compromised, the CTL file can be signed by SAST2 to prevent the phones from being reset to their factory defaults.

Examples

The following example names sast1tp as the SAST1 trustpoint.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

Related Commands

Command	Description
sast2 trustpoint	Specifies the name of the SAST2 trustpoint.

sast2 trustpoint

To specify the name of the SAST2 trustpoint, use the **sast2 trustpoint** command in CTL-client configuration mode. To return to the default, use the **no** form of this command.

sast2 trustpoint *label*
no sast2

Syntax Description

label	Name of the SAST2 trustpoint.

Command Default

No SAST2 trustpoint name is specified.

Command Modes

CTL-client configuration (config-ctl-client)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

The **sast1 trustpoint** and **sast2 trustpoint** commands are used to set up the System Administrator Security Token (SAST) credentials, which are used to sign the CTL file. The SAST1 and SAST2 certificates must be different from each other, but to conserve memory either one of them can use the same certificate as Cisco CME. The CTL file is always signed by SAST1 credentials. The SAST2 credentials are included in the CTL file so that if the SAST1 certificate is compromised, the CTL file can be signed by SAST2 to prevent the phones from being reset to their factory defaults.

Examples

The following example names sast2tp as the SAST2 trustpoint.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

Related Commands

Command	Description
sast1 trustpoint	Specifies the name of the SAST1 trustpoint.

sdspfarm conference lecture mode on

To permit a participant in a video conference call to switch back and forth between lecture mode and the the configured default mode in DSP farm, use the **sdspfarm conference** command in telephony-service configuration mode. The participant who enters the FAC becomes the lecturer and is displayed on all other screens. The lecturer's screen displays a scanning stream of the other participants.

To delete a tag generated by the **sdspfarm conference** command, use the **no** form of this command.

sdspfarm conference lecture mode on *FAC* **release** *FAC*
no sdspfarm conference lecture mode on *FAC* **release** *FAC*

Syntax Description

FAC	Sets the Feature Access Codes (FAC) that a participants enters on the keypad to switch to the lecture mode. Valid values are the numbers on the keypad. Maximum 3 digits
release *FAC*	Sets the Feature Access Codes (FAC) that a participants enters on the keypad to exit lecture mode. Valid values are the numbers on the keypad. Maximum 3 digits

Command Default

Lecture mode is not enabled by default.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.

Usage Guidelines

You can define any three digits to be FAC for lecture mode. A participant cannot enter lecture mode on a phone with unsupported video formats, for example an audio-only phone. The lecture mode participant must exit lecture mode before anyone else can become the lecturer.

Examples

The following example configure lecture mode to be activated when the user presses a FAC number of 111.

```
Router(config)# telephony-service
outer(config-telephony)# sdspfarm conference lecture-mode on 111 release 222
```

Related Commands

Command	Description
dspfarm profile	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.
sdspfarm transcode	Specifies the maximum number of transcoding sessions allowed per Cisco CME router.
sdspfarm units	Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.

sdspfarm conference mute-on mute-off

To define mute-on and mute-off DTMF digits for use during conferencing, use the **sdspfarm conference mute-on mute-off** command in telephony-service configuration mode. To disable the mute-on and mute-off digits, use the **no** form of this command.

sdspfarm conference mute-on *mute-on-digits* **mute-off** *mute-off-digits*
no sdspfarm conference mute-on *mute-on-digits* **mute-off** *mute-off-digits*

Syntax Description

mute-on *mute-on-digits*	Defines the buttons you press on your phone to mute during a conference. Maximum: 3 digits. Valid values are the numbers and symbols that appear on your telephone keypad: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, and #.
mute-off *mute-off-digits*	Defines the buttons you press on your IP phone to unmute during a conference. Maximum: 3 digits. Valid values are the numbers and symbols that appear on your telephone keypad: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, and #.

Command Default

No mute-on or mute-off digits are defined.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ2	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

You must define mute-on and mute-off digits to mute or unmute your phone using the keypad during a conference. The mute-on digits and mute-off digits can be the same or different. You can mute and unmute your phone using the phone's mute button also. You must unmute the phone in the same way that you muted it, either with the keypad or the mute button.

Examples

The following example configures #5 as the buttons to press to mute and unmute the phone during a conference:

```
Router(config-telephony)# sdspfarm conference mute-on #5 mute-off #5
```

sdspfarm tag

To permit a digital-signal-processor (DSP) farm to be to registered to Cisco Unified CME and associate it with the MAC address of a Skinny Client Control Protocol (SCCP) interface, use the **sdspfarm tag** command in telephony-service configuration mode. To delete a tag generated by the **sdspfarm tag** command, use the **no** form of this command.

sdspfarm tag *number device-name*
no sdspfarm tag *number device-name*

Syntax Description

number	Numeric name for a DSP farm. Number from 1 to 10.
device-name	Word describing the device, such as the MAC address, of the SCCP client interface that is preceded by the Message Transfer Part (MTP).

Command Default

DSP farm is not created.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco Unified CME 3.2	This command was introduced.
15.1(4)M	Cisco Unified CME 8.6	Increased support for the number of DSP farms to 10.

Usage Guidelines

DSP farm profiles are sets of DSP resources used for conferencing and transcoding only. DSP farms do not include voice termination resources. Use the **show interface** command to find the MAC address of the SCCP client interface.

Examples

The following example declares tag 1 as the MAC address of mac000a.8aea.ca80. The **show interface** command is used to obtain the MAC address.

```
Router# show interface FastEthernet 0/0
.
.
.
FastEthernet0/0 is up, line protocol is up
Hardware is AmdFE, address is 000a.8aea.ca80 (bia 000a.8aea.ca80)
.
.
.
Router(config)# telephony-service

Router(config-telephony)# sdspfarm tag 1 mac000a.8aea.ca80
```

Related Commands

Command	Description
sdspfarm transcode	Specifies the maximum number of transcoding sessions allowed per Cisco CME router.

Command	Description
sdspfarm units	Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.

sdspfarm transcode sessions

To specify the maximum number of transcoding sessions allowed per Cisco CallManager Express (Cisco CME) router, use the **sdspfarm transcode sessions** command in telephony-service configuration mode. To return to the default transcode session of 0, use the **no** form of this command.

sdspfarm transcode sessions *number*
no sdspfarm transcode sessions *number*

Syntax Description

number	Declares the number of DSP farm sessions. Valid values are numbers from 1 to 128.

Command Default

The default is 0.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco Unified CME 3.2	This command was introduced.

Usage Guidelines

The transcoding is allowed between G.711 and G.729. A session consists of two transcode streams. To configure this information, you must know how many digital-signal-processor (DSP) farms are configured on the network module (NM) farms in your Cisco CME router. DSP farms are sets of DSP resources used for conferencing and transcoding only. DSP farms do not include voice termination resources. To learn how many DSP farms have been configured on your Cisco CME router, use the **show sdspfarm** command.

Examples

The following example sets the maximum number of transcoding sessions allowed on the Cisco CME router to 20:

```
Router(config)# telephony-service

Router(config-telephony)# sdspfarm transcode sessions 20
```

Related Commands

Command	Description
sdspfarm tag	Declares a DSP farm and associates it with an SCCP client interface's MAC address.
sdspfarm unit	Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.
show sdspfarm	Displays the status of the configured DSP farms and transcoding streams.

sdspfarm units

To specify the maximum number of digital-signal-processor (DSP) farm profiles that are allowed to be registered to the Skinny Client Control Protocol (SCCP) server, use the **sdspfarm units** command in telephony-service configuration mode. To set the number of DSP farm profiles to the default value of 0, use the **no** form of this command.

sdspfarm units *number*
no sdspfarm units *number*

Syntax Description

number	Number of DSP farms. Valid values are numbers from 0 to 10.

Command Default

The default number is 0.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco Unified CME 3.2	This command was introduced.
15.1(4)M	Cisco Unified CME 8.6	Increased support for the number of DSP farms to 10.

Usage Guidelines

DSP farm profiles are sets of DSP resources used for conferencing and transcoding only. DSP farm profiles do not include voice termination resources.

Examples

The following example configures a Cisco CME router to register one DSP farm:

```
Router(config)# telephony-service
Router(config-telephony)# sdspfarm units 1
```

Related Commands

Command	Description
sdspfarm tag	Declares a DSP farm and associates it with the MAC address of an SCCP client interface.
sdspfarm transcode	Specifies the maximum number of transcoding sessions allowed per Cisco CME router.

sdspfarm unregister force

To remove all transcoding streams associated with active calls, use the **sdspfarm unregister force** command in telephony-service configuration mode. To deactivate the removal of transcoding streams, use the **no** form of this command.

sdspfarm unregister force
no sdspfarm unregister force

Syntax Description

This command has no arguments or keywords.

Command Default

The default is transcoding streams are not removed.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco product	Modification
12.3(11)T	Cisco Unified CME 3.2	This command was introduced.

Usage Guidelines

If any of the SCCP server's associated streams are functioning in active calls, the default response for the **sdspfarm unregister force** command is to reject them. If no stream is used in a call, all of the transcoding streams associated with the DSP farm will be released, and SCCP server can recycle those streams for other DSP farms.

Examples

The following example removes all transcoding streams for active calls.

```
Router(config)# telephony-service
Router(config-telephony)# sdspfarm unregister force
```

Related Commands

Command	Description
sdspfarm tag	Declares a DSP farm and associates it with an SCCP client interface's MAC address.
sdspfarm unit	Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.
show sdspfarm	Displays the status of the configured DSP farms and transcoding streams.

secondary dialtone (voice port)

To allow dialed digits to be collected from the remote switch when “connection plar” is not defined from the analog FXO voice-port, use the secondary dialtone command in global configuration mode. To disable the secondary dialtone, use the no form of the command.

secondary dialtone
no secondary dialtone

Syntax Description

This command has no arguments or keywords.

Command Default

The secondary dialtone command is disabled.

Command Modes

Global configuration.

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use the secondary dialtone command to allow dialed digits to be collected from the remote switch when “connection plar” is not defined from the analog FXO voice-port.

The following is a sample output from this command:

```
Router(config)# voice-port 2/0/0
Router (config-voiceport)#no secondary ?
  dialtone  Secondary dialtone option for FXO port
Router (config-voiceport)#no secondary dialtone
"secondary dialtone" is used to enable 2-stage dialing for an
incoming call
!
```

Related Commands

Command	Description
voice service	Enters voice service configuration mode.

secondary start

To specify the ephone hunt group agent to receive parked calls that are forwarded to the secondary pilot number, use the **secondary start** command in ephone-hunt configuration mode. To disable this setting, use the **no** form of this command.

secondary start [{**current** | **next***list-position*}]
no secondary start [{**current** | **next***list-position*}]

Syntax Description

current	The ephone-dn that parked this call.
next	The ephone-dn that follows the parking ephone-dn in the list specified by the **list** command.
list-position	The ephone-dn at the specified position in the list specified by the **list** command. Range is from 1 to 20.

Command Default

No hunt-group agent is specified for receiving parked calls that are forwarded to the secondary pilot number.

Command Modes

Ephone-hunt configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

When a call that has been parked by a hunt group agent meets either of these conditions, the hunt group agent to receive the call can be specified with the **secondary start** command:

- The call is recalled from call park to the hunt group secondary pilot number.
- The call is transferred from call park to an ephone-dn that forwards the call to the hunt group secondary pilot number.

Examples

The following example specifies that the third hunt group member (3031) will receive calls that are recalled or forwarded to the secondary group hunt pilot number (3001) after the calls have been parked by an ephone-dn.

```
ephone-hunt 17 sequential
 pilot 3000 secondary 3001
 list 3011, 3021, 3031
 timeout 10
 final 7600
 secondary start 3
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.

Command	Description
list	Creates a list of extensions that are members of an ephone hunt group

secondary-dialtone

To activate a secondary dial tone when a Cisco IP phone user dials a defined public switched telephone network (PSTN) access prefix, use the **secondary-dialtone** command in telephony-service configuration mode. To disable the secondary dial tone, use the **no** form of this command.

secondary-dialtone *digit-string*
no secondary-dialtone

Syntax Description

digit-string	String of up to 32 numbers that defines an access prefix.

Command Default

No secondary dial tone is enabled.

Command Modes

Telephony-service configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The secondary dial tone is turned off when the next number after the access prefix is pressed. For example, if 8 is the access prefix and a person dials 8 555-0145, the secondary dial tone is turned off when the digit 5 is pressed.

Note The symbol # is considered to be the terminating string of a dial string. Hence, it is not supported under **secondary-dialtone**, to avoid conflict with dial-peer matching.

Examples

The following example enables a secondary dial tone when a Cisco IP phone users press the digit 9 to get an outside line:

```
Router(config)# telephony-service
Router(config-telephony)# secondary-dialtone 9
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

secure-signaling trustpoint

To specify the name of the PKI trustpoint with the certificate to use for TLS handshakes with IP phones on TCP port 2443, use the **secure-signaling trustpoint** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

secure-signaling trustpoint *label*
no secure-signaling trustpoint

Syntax Description

label	Name of a configured PKI trustpoint with a valid certificate.

Command Default

No trustpoint is specified.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to name the trustpoint that enables handshaking between Cisco Unified CME and a phone to ensure secure SCCP signaling on TCP port 2443.

Examples

The following example names server25, the CAPF server, as the trustpoint to enable secure SCCP signaling:

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authenticated
Router(config-telephony)# secure-signaling trustpoint server25
Router(config-telephony)# tftp-server-credentials trustpoint server12
Router(config-telephony)# load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
Router(config-telephony)# exit
```

semi-attended enable (voice register template)

To enable call transfer at the alert call stage for supported SIP phones in Cisco Unified CME, use the **semi-attended enable** command in the voice register template mode. To disable call transfer, use the **no** form of this command.

semi-attended enable
no semi-attended enable

Syntax Description

This command has no arguments or keywords.

Command Default

Call transfer at the alert call stage is enabled.

Command Modes

Voice register template (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command enables a call transfer at the alert stage in the specified template which can then be applied to SIP phones in Cisco Unified CME. Semi-attended call transfer is enabled by default. To disable semi-attended call transfer, use the **no semi-attended** command.

To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following partial output from the **show-running config** command shows that the semi-attended call transfer is disabled in voice register template 1:

```
Router# show running-config
!
.
.
.
!
voice register template 1
 no semi-attended enabled
!

The following example shows how to enable semi-attended call transfer in a template:
Router(config)# voice register template 1
Router(config-register-temp)# semi-attend enable
```

Related Commands

Command	Description
template (voice register pool)	Applies template to SIP IP phone being configured.

server (CTL-client)

To enter trustpoint information for the CAPF server, Cisco Unified CME router, or TFTP server into the router configuration, use the **server** command in CTL-client configuration mode. To return to the default, use the **no** form of this command.

server {**capf** | **cme** | **cme-tftp** | **tftp**} *ip-address* **trustpoint** *label*
no server {**capf** | **cme** | **cme-tftp** | **tftp**} *ip-address*

Syntax Description

capf	CAPF server.
cme	Cisco Unified CME router.
cme-tftp	Combined Cisco Unified CME router and TFTP server.
tftp	TFTP server.
ip-address	IP address of the entity.
trustpoint *label*	Name of the PKI trustpoint for the entity.

Command Default

Trustpoint information about the CAPF server, Cisco Unified CME router, or TFTP server is not present in the security configuration.

Command Modes

CTL-client configuration (config-ctl-client)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication. Cisco IOS software stores credential information in a trustpoint. The trustpoint label in this command names the specified PKI trustpoint.

Note Repeat this command for each entity that requires a trustpoint.

Examples

The following example defines trustpoint names and IP addresses for the CAPF server, the Cisco Unified CME router, and the TFTP server:

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
```

```
Router(config-ctl-client)# regenerate
```

server (presence)

To specify the IP address of a presence server for sending presence requests from internal watchers to external presence entities, use the **server** command in presence configuration mode. To remove the server, use the **no** form of this command.

server *ip-address*
no server

Syntax Description

ip-address	IP address of the remote presence server.

Command Default

A remote presence server is not used.

Command Modes

Presence configuration (config-presence)

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command specifies the IP address of a presence server that handles presence requests when the watcher and presence entity (presentity) are not collocated. The router acts as the presence server and processes all presence requests and status notifications when a watcher and presentity are both internal. If a subscription request is for an external presentity, the request is sent to the remote server specified by this command.

Examples

The following example shows a presence server with IP address 10.10.10.1:

```
Router(config)# presence
Router(config-presence)# allow subscribe
Router(config-presence)# server 10.10.10.1
```

Related Commands

Command	Description
allow subscribe	Allows internal watchers to monitor external presence entities (directory numbers).
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
max-subscription	Sets the maximum number of concurrent watch sessions that are allowed.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.
watcher all	Allows external watchers to monitor internal presence entities (directory numbers).

server-security-mode

To change the security mode of the Cisco Unified CME phone authentication server, use the **server-security-mode** command in telephony-service configuration mode. To change the mode from secure to nonsecure, use the **no** form of this command.

server-security-mode {**erase** | **non-secure** | **secure**}
no server-security-mode

Syntax Description

erase	Deletes the certificate trust list (CTL) file.
non-secure	Enables nonsecure mode.
secure	Secure mode.

Command Default

When the CTL file is initially generated, the mode is set to secure.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(22)T	Cisco Unified CME 7.0	The **erase** keyword was added.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

This command has no effect until the CTL file is initially generated by the CTL client. When the CTL file is successfully generated, the CTL client automatically sets the server security mode to secure. You can then toggle the mode from secure to nonsecure using this command.

After toggling between secure and non-secure mode, you must use the **regenerate** command in CTL-client configuration mode to generate the CTL file. This is necessary because if the security mode is nonsecure, its credentials are zeroed out in the CTL file. If the security mode is secure, the CTL file contains the server's credentials.

The **no** version of this command sets the mode to non-secure; it does not remove the command from your configuration.

To remove this command from your configuration and revert to the state before the Cisco Unified CME security feature was activated, use the **erase** keyword and follow the instructions displayed on the console. When you use this command with the **erase** keyword, the router checks whether the Cisco IOS CTL-provider process is running, and if not, it deletes the CTL file from router storage. After using this command to delete the CTL file, you must manually delete the CTL file from any SCCP phones that had downloaded it previously.

Examples

The following example changes the mode of the server to non-secure.

```
telephony-service
```

```
server-security-mode non-secure
```

Related Commands

Command	Description
regenerate	Creates a new CTLFile.tlv file after changes are made to the CTL client configuration.

service directed-pickup

To enable Directed Call Pickup and modify the function of the GPickUp and PickUp soft keys, use the **service directed-pickup** command in telephony-service configuration mode. To disable Directed Call Pickup, use the **no** form of this command.

service directed-pickup **[gpickup]**
noservice directed-pickup **[gpickup]**

Syntax Description

gpickup	(Optional) Enables phone users to perform Directed Call Pickup using the GPickUp soft key.

Command Default

For SCCP phones, Directed Call Pickup using the PickUp soft key is enabled.

For SIP phones, Directed Call Pickup is not enabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(22)YB	Cisco Unified CME 7.1	The **gpickup** keyword and support for SIP phones was added.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command modifies the function of the GPickUp and PickUp soft keys for the Directed Call Pickup and Local Group Pickup features.

To globally disable Directed Call Pickup on all phones, use the no form of this command. The no form of this command also changes the behavior of the PickUp soft key on IP phones so that a user pressing it invokes Local Group Pickup instead of Directed Call Pickup.

To selectively remove the PickUp soft key from one or more SCCP phones, use the **features blocked** command in ephone-template mode. The **features blocked** command removes the PickUp soft key from SCCP IP phones and blocks Directed Call Pickup on analog phones to which you apply the template.

The table describes the use of the GPickUp and PickUp soft keys for each feature depending on the setting of this command.

Task ID

Table 14: service directed-pickup Command Comparison

Cisco IOS Command Syntax SIP Phones service directed-pickup gpickup	SCCP Phones	SIP Phones
Directed Call Pickup (Call on any ringing extension)	GPickUp soft key and extension	
Local Group Pickup (Call in same group)	GPickUp soft key and * or PickUp soft key	
Other Group Pickup (Call in different group)	GPickUp soft key and pickup group number	
service directed-pickup (default)		
Directed Call Pickup	PickUp soft key and extension	—
Local Group Pickup	GPickUp soft key and *	GPickUp soft key and * or Pickup soft key
Other Group Pickup	GPickUp soft key and pickup group number	
no service directed-pickup		
Directed Call Pickup		—
Local Group Pickup		GPickUp soft key and * or PickUp soft key
Other Group Pickup		GPickUp soft key and pickup group number

[1]

Example

The following example shows that Directed Call Pickup is disabled globally:

```
telephony-service
no service directed-pickup
```

The following example shows that Directed Call Pickup, Group Pickup, and Local Group Pickup can be performed using the GPickUp soft key:

[1] Supported in Cisco Unified CME 7.1 and later versions.

```
telephony-service
service directed-pickup gpickup
```

Related Commands

Command	Description
call-feature-uri	Creates a new CTLFile.tlv file after changes are made to the CTL client configuration. Specifies the uniform resource identifier (URI) for soft keys on SIP phones registered to Cisco Unified CME.
features blocked	Prevents one or more features from being used on SCCP phones.
pickup-group	Assigns an extension to a call-pickup group.

service dnis dir-lookup

To allow the display of names associated with called numbers for incoming calls on IP phones, use the **service dnis dir-lookup** command in telephony-service configuration mode. To deactivate directory lookup, use the **no** form of this command.

service dnis dir-lookup
no service dnis dir-lookup

Command Default

The default is directory service lookup is inactive.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.

Usage Guidelines

The **service dnis dir-lookup** command provides a called number to the name-lookup service to support display of the name associated with the called number for incoming calls to IP phones. The display name is obtained from the Cisco CME system's list of Cisco CME directory names created using the **directory entry** command in the telephony-service configuration mode.

Called numbers can be displayed for overlaid ephone-dn and for ephone-dns that are not overlaid. Secondary line are supported.

To allow a single ephone-dn to receive calls for multiple different called numbers (with different names), you must use wildcard "." characters in the number field for the ephone-dn.

To use the **service dnis dir-lookup** command in conjunction with the **ephone-hunt**, you can configure the ephone-hunt group to use a pilot number that contains wildcard "." characters. This command allows the ephone-hunt group to receive calls from different numbers. Individual ephone-dns that are configured as members of the hunt group with the **ephone-hunt list** must not have wildcard characters in their number fields.

If the **service dnis dir-lookup** is used at the same time as the **service dnis overlay**, the directory-lookup service takes precedence in resolving the name for the called number.

Examples

The following is an example of an overlaid ephone-dn configuration, where the **service dnis dir-lookup** allows one of three directory entry names to be displayed on three IP phones when a call is placed to a number declared in the **directory entry command.**

```
telephony-service
 service dnis dir-lookup
 directory entry 1 0001 name dept1
 directory entry 2 0002 name dept2
 directory entry 3 0003 name dept3
ephone-dn 1
 number 0001
ephone-dn 2
 number 0002
ephone-dn 2
 number 0002
ephone 1
```

```
 button 1o1,2,3
ephone 2
 button 1o1,2,3
ephone 3
 button 1o1,2,3
```

The following is an example of an ephone-dn configuration in which the overlay function is not in use. There are three IP phones, each with two buttons. Button 1 receives calls from user1, user2, and user3; button 2 receives calls from user4, user5, and user6.

```
telephony-service
 service dnis dir-lookup
 directory entry 1 5550001 name user1
 directory entry 2 5550002 name user2
 directory entry 3 5550003 name user3
 directory entry 4 5550010 name user4
 directory entry 5 5550011 name user5
 directory entry 6 5550012 name user6
ephone-dn 1
 number 555000.
ephone-dn 2
 number 5552001.
ephone 1
 button 1:1
 button 2:2
 mac-address 1111.1111.1111
ephone 2
 button 1:1
 button 2:2
 mac-address 2222.2222.2222
```

The following is an example of a hunt-group configuration. There are three phones, each with two buttons, and each button is assigned two numbers. When a person calls 5550341, Cisco CME matches the hunt-group pilot secondary number (555....) and rings button 1 on one of the two phones and displays “user1.” The selection of the phone is dependent on the **ephone-hunt** settings.

```
telephony-service
 service dnis dir-lookup
 max-redirect 20
 directory entry 1 5550341 name user1
 directory entry 2 5550772 name user2
 directory entry 3 5550263 name user3
 directory entry 4 5550150 name user4
ephone-dn 1
 number 1001
ephone-dn 2
 number 1002
ephone-dn 3
 number 1003
ephone-dn 4
 number 1004
ephone 1
 button 1o1,2
 button 2o3,4
 mac-address 1111.1111.1111
ephone 2
 button 1o1,2
 button 2o3,4
 mac-address 2222.2222.2222
ephone-hunt 1 peer
 pilot 1000 secondary 555....
 list 1001, 1002, 1003, 1004
```

```
 final 5556000
 hops 5
 preference 1
 timeout 20
 no-reg
```

The following is an example of a secondary-line configuration. Ephone-dn 1 can accept calls from extension 1001 and from 5550000, 5550001, and 5550002.

```
telephony-service
 service dnis dir-lookup
 directory entry 1 5550000 name doctor1
 directory entry 2 5550001 name doctor2
 directory entry 3 5550002 name doctor3
ephone-dn 1
 number 1001 secondary 555000.
ephone 1
 button 1:1
 mac-address 2222.2222.2222
```

Related Commands

Command	Description
directory entry	Adds an entry to a local phone directory that can be displayed on IP phones.
ephone-hunt	Enters ephone-hunt configuration mode to create a hunt group for use in a Cisco CME system.
list	Creates a list of extensions that are members of a Cisco CME ephone hunt group.
service dnis overlay	Allows an ephone-dn name to appear on receiving IP phones' displays when the ephone-dn's number is called.

service dnis overlay

To allow incoming calls to an ephone-dn overlay to display called ephone-dn names, use the **service dnis overlay** command in telephony-service configuration mode. To deactivate the service dialed number identification service (DNIS) overlay, use the **no** form of this command.

service dnis overlay
no service dnis overlay

Command Default

The ephone-dn names in an ephone-dn overlay are not displayed on IP phones.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	Cisco Unified CME 3.2	This command was introduced.

Usage Guidelines

The **service dnis overlay** allows phone users to determine which ephone-dn within an overlay set is being called. Up to ten ephone-dns are allowed per overlay set. When an incoming call is presented under a **service dnis overlay** configuration, the phone displays the name of the individual ephone-dn according to the **name** configured under the ephone-dn configuration mode. Note that for an ephone-dn name to be displayed on IP phones, you must first assign ephone-dn names with the **name** command in ephone-dn configuration mode.

The number of the first ephone-dn listed in the **button** is the default display for all phones using the same set of overlaid ephone-dns. Calls to the first ephone-dn display the caller ID. Calls to the remaining ephone-dns display ephone-dn names. For example, if there are three phones with one overlay set containing five ephone-dns, the first ephone-dn's number listed is the default display for all three phones. A call to the first ephone-dn displays the caller ID on all phones until the call is picked up. When the call is answered by phone 1, the displays in phone 2 and phone 3 return to the default display. Calls to the last four ephone-dns display ephone-dn names.

If the **service dnis overlay** is used at the same time as the **service dnis dir-lookup**, the **service dnis dir-lookup** takes precedence in determining the name to be displayed.

Examples

The following is an overlay configuration for two phones with button 1 assigned to pick up three 800 numbers from three ephone-dns that have been assigned names. The default display for button 1 is 18005550100. A call to 18005550100 displays the caller ID. Calls to 18005550001 and 18005550002 display "name1" and "name2," respectively.

```
telephony-service
 service dnis overlay
ephone-dn 1
 name mainnumber
 number 18005550100
ephone-dn 2
 name name1
 number 18005550001
ephone-dn 3
 name name2
 number 18005550002
ephone 1
 button 1o1,2,3
```

```
ephone 2
 button 1o1,2,3
```

Related Commands

Command	Description
name	Associates a name with a Cisco CME extension (ephone-dn).
service dnis dir-lookup	Allows directory entry lookup for the display of directory entry names on IP phones.

service dss

To enable DSS (Direct Station Select) in a Cisco Unified CME system, use the **service dss** command in telephony-service configuration mode. To globally disable the DSS feature, use the **no** form of this command.

service dss
no service dss

Syntax Description

This command has no arguments or keywords.

Command Default

DSS service is disabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(6)XE	Cisco Unified CME 4.0(2)	This command was introduced.
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)T	Cisco Unified CME 4.0(3)	This command is integrated into Cisco IOS Release 12.4(11)T.

Usage Guidelines

This command enables phone users to quickly transfer calls to an extension selected by a speed-dial or monitor line button without having to press the Transfer button. If this command is enabled, a user can transfer a call when the call is in the connected state by simply pressing a speed-dial or monitor line button to select the transfer destination. The transfer action is automatically implied by CME if the **service dss** is enabled.

This feature is supported only on phones on which monitor-line buttons for speed dial or speed-dial line buttons are configured.

Using the **no** form of the changes the behavior of the speed-dial line button on all IP phones so that a user pressing a speed-dial button in the middle of a connected call will play out the speed-dial digits into the call without transferring the call. If the **service dss** is disabled, the phone user must press the Transfer button before pressing the speed-dial line button or monitor line button to transfer the call.

For Cisco Unified CME 4.0 and earlier, the **transfer-system full-consult dss** is used to select between blind transfers and consult transfers for the DSS case.

Examples

The following example globally enables directed call pickup.

```
telephony-service
 service dss
```

Related Commands

Command	Description
button	Associates ephone-dns with individual buttons on a Cisco Unified IP phone and to specify line type, such as monitor mode for a shared line.

Command	Description
speed-dial	Defines a unique speed-dial identifier, a digit string to dial, and an optional label to display next to a line button.

service https (ephone-template)

To locally provision Hypertext Transfer Protocol Secure (HTTPS) services access from Cisco Unified SCCP IP phones on Cisco Unified CME, use the **service https** command in ephone-template configuration mode. To disable access to HTTPS services, use the **no** form of this command.

service https
no service https

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco Unified SCCP IP phones are unable to access HTTPS services on Cisco Unified CME.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

Use the **service https** command to enable access to HTTPS services like local-directory lookup, My Phone Apps, and Extension Mobility.

Examples

The following example shows how to locally provision HTTPS services from Cisco Unified SCCP IP phones:

```
configure terminal
ephone-template 1
     service https
```

Command	Description
ephone-template	Enters ephone-template configuration mode and creates an ephone template to configure a set of phone features.

service https (telephony-service)

To globally provision Hypertext Transfer Protocol Secure (HTTPS) services access from Cisco Unified SCCP IP phones on Cisco Unified CME, use the **service https** command in telephony-service configuration mode. To disable accesss to HTTPS services, use the **no** form of this command.

service https
no service https

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco Unified SCCP IP phones are unable to access HTTPS services on Cisco Unified CME.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

Use the **service https** command to enable access to HTTPS services like local-directory lookup, My Phone Apps, and Extension Mobility.

Examples

The following example shows how to globally provision HTTPS services from Cisco Unified SCCP IP phones:

```
configure terminal
telephony-service
     cnf-file perphone
     service https
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

service https (voice register global)

To globally provision Hypertext Transfer Protocol Secure (HTTPS) services access from Cisco Unified SIP IP phones on Cisco Unified CME, use the **service https** command in voice register global configuration mode. To disable access to HTTPS services, use the **no** form of this command.

service https
no service https

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco Unified SIP IP phones are unable to access HTTPS services on Cisco Unified CME.

Command Modes

Voice register global configuration (config-register-global)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

Use the **service https** command to enable access to HTTPS services like local-directory lookup, My Phone Apps, and Extension Mobility.

Examples

The following example shows how to globally provision HTTPS services from Cisco Unified SIP IP phones:

```
configure terminal
voice register global
     service https
```

Related Commands

Command	Description
voice register global	Enters voice register global configuration mode and sets global parameters for all supported Cisco Unified SIP IP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.

service https (voice register template)

To locally provision Hypertext Transfer Protocol Secure (HTTPS) services access from Cisco Unified SIP IP phones on Cisco Unified CME, use the **service https** command in voice register template configuration mode. To disable access to HTTPS services, use the **no** form of this command.

service https
no service https

Syntax Description

This command has no arguments or keywords.

Command Default

Cisco Unified SIP IP phones are unable to access HTTPS services on Cisco Unified CME.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

Use the **service https** command to enable access to HTTPS services like local-directory lookup, My Phone Apps, and Extension Mobility.

Examples

The following example shows how to globally provision HTTPS services from Cisco Unified SIP IP phones:

```
configure terminal
voice register template 1
     service https
```

Related Commands

Command	Description
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

service local-directory

To enable the availability of the local directory service on IP phones served by the Cisco Unified Communications Manager Express (Unified CME) router, use the **service local-directory** command in telephony service configuration mode. To disable the display, use the **no** form of this command.

service local-directory [**authenticate**][**username**][**password**] **[0|6]** *password*
no service local-directory [**authenticate**][**username**][**password**]

Syntax Description

authenticate	(Optional) Requires authentication for local directory search requests.
username	(Optional) Provides username for authentication of local directory server.
password [0|6]	(Optional) Provides password for authentication of local directory server. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Command Default

Local directory service is available on IP Phones.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modifications
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The **authenticate** keyword was introduced.
12.3(4)	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
Cisco IOS XE Everest 16.6.1	Unified CME 12.0	This command was enhanced to authenticate the username and password for accessing the local directory service.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption, based on Unified CME password policy.

Usage Guidelines

Use this command with Cisco IOS Telephony Services V2.1, Cisco CME 3.0, or a later version.

When you configure the **url directories** command with the URL and credentials of the server that hosts the local directory, the command takes precedence over **service local-directory[authenticate] [username][password]**. When you configure the **url directories** command with only the URL of the server that hosts the local directory, Unified CME tries to fetch the username and password credentials from the command **service local-directory[authenticate] [username][password]**, if it is configured.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]**. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Example 1

The following example specifies that the directory service should not be available on the IP phones served by this ITS router:

```
Router(config)# telephony-service
Router(config-telephony-service)# no service local-directory
```

Example 2

The following example configures the username and password for accessing the server that hosts the directory service. In this scenario, the command **url directories** is not configured.

```
Router(config)# telephony-service
Router(config-telephony-service)# service local-directory authenticate admin cisco12345
```

The output for this sample configuration in the CNF XML file is as follows:

```
<directoryURL>http://admin:cisco12345/8.39.16.26:80/localdirectory</directoryURL>
```

Example 3

The following example specifies the configuration when the command **service local-directory[authenticate] [username][password]** is configured and the command **url directories** is configured without credentials. In this scenario, the server URL is updated with the credentials provided in the **service local-directory** CLI command.

```
Router(config)# telephony-service
Router(config-telephony-service)# service local-directory authenticate root cisco
Router(config-telephony-service)#  url directories http://8.39.16.26:80/localdirectory
```

The output for this sample configuration in the CNF XML file is as follows:

```
<directoryURL>http://root:cisco/8.39.16.26:80/localdirectory</directoryURL>
```

Example 4

The following example specifies the configuration when the CLI commands **url directories** and **service local-directory[authenticate] [username][password]** are configured with credentials. In this scenario, the server URL is updated with the credentials provided in the **url directories** CLI command.

```
Router(config)# telephony-service
Router(config-telephony-service)# service local-directory authenticate admin cisco
Router(config-telephony-service)#  url directories
http://root:cisco123@8.39.16.26:80/localdirectory
```

The output for this sample configuration in the CNF XML file is as follows:

```
<directoryURL>http://root:cisco123@8.39.16.26:80/localdirectory</directoryURL>
```

Example 5

The following example specifies the configuration when the CLI command **service local-directory** is configured and the commands **url directories** and **service local-directory[authenticate] [username][password]** are not configured. In this scenario, the local directory service is activated though the credentials are not configured. Hence, the XML files generated by tftp-bindings will contain only the URL information of the server without the username and password credentials.

```
Router(config)# telephony-service
Router(config-telephony-service)# service local-directory
```

The output for this sample configuration in the CNF XML file is as follows:

```
Router# show telephony-service tftp-bindings
more flash:/its/vrf1/SEP5057A88797E0.cnf.xml
 <directoryURL>http://8.39.16.26:80/localdirectory</directoryURL>
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

service phone

To modify the vendorConfig parameters in the configuration file, use the **service phone** command in telephony-service or ephone-template configuration mode. To disable a setting, use the **no** form of this command.

service phone *parameter-name parameter-value*
no service phone *parameter-name parameter-value*

Syntax Description

parameter-name	Name of the vendorConfig parameter in the configuration file. For valid parameter names, see the table below. Parameter names are word and case-sensitive and must be entered exactly as shown.
parameter-value	Value for the vendorConfig parameter. For valid values and defaults, see the table below.

Command Default

The vendorConfig parameters in phone configuration files are set to default values.

Command Modes

Telephony-service configuration (config-telephony)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode for certain parameters.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.
15.1(4)M	Cisco Unified CME 8.6	This command was modified. The xml config file argument was added.
Cisco IOS XE Fuji 16.9.1	Unified CME 12.3	This command **service phone lineMode 1** introduces support for Enhanced Line Mode (ELM) for Cisco IP Phone 8800 Series on Unified CME.

Usage Guidelines

This command in telephony-service configuration mode modifies vendorConfig parameters in configuration file for phones in a Cisco Unified CME system.

The vendorConfig section of a configuration file is read by a phone’s firmware when that Cisco Unified IP phone is booted. The number and type of parameters may vary from one firmware version to the next.

If a firmware version does not support a particular parameter, that parameter cannot be implemented. For example, Cisco phone firmware 8.2.1 or a later version is required to support the G.722-64K codec on G.722-capable SCCP phones and Cisco phone firmware 8.3.1 or a later version is required to support the G.722-64K codec on G.722-capable SIP phones. If your phones are loaded with an earlier version of phone firmware, they cannot support the G.722-64K codec regardless of how the **g722CodecSupport** parameter is configured.

The IP phone that downloads the configuration file will implement only those parameters that it can support and ignore configured parameters that it cannot implement. For example, a Cisco IP phone without a backlit display cannot implement backlight parameters regardless of how they are configured.

In Cisco Unified CME 4.0 and later versions, support for creating configuration files at a phone level was added for SCCP phones. This command in ephone-template configuration mode creates an template of vendorConfig parameters that can be applied to individual SCCP phones in Cisco Unified CME. This command in ephone-template configuration mode does not work for all vendorConfig parameters. See the table below for information about individual parameters.

In Cisco Unified CME 4.0 and later versions, if you use an ephone template to apply this command to one or more phones, you must also configure the **cnf-file perphone** command so that a separate configuration file is created for each phone, by MAC address. To apply this command in telephony-service mode to all phones of a particular type in Cisco Unified CME 4.0 and later versions, you can configure the **cnf-file perphonetype** command to specify that configuration files are generated by phone type.

To apply this command in telephony-service configuration mode to all phones in your Cisco Unified CME system, ensure that the system is configured for the default single per-system configuration file for all phones.

If you use an ephone template to apply this command to a phone and you also use this command in telephony-service configuration mode for the same phone, the value that you set in ephone-template configuration mode has priority.

After modifying the vendorConfig parameters, you must generate new configuration files.

After generating configuration files, reset or reboot the IP phone to be configured to download the new configuration file.

From Unified CME Release 12.3, you can enable Enhanced Line Mode on Unified for Cisco IP Phone 8800 Series (except 8821, 8831, 8832 models) by configuring the CLI command **service phone lineMode 1** under **telephony-service** configuration mode. The Cisco IP Phone 8800 Series configured on Unified CME uses the vendor config XML body in the CNF file to verify if the CLI command **service phone lineMode 1** is added to enable ELM mode. By default, ELM is not enabled on Unified CME. To disable ELM on the Unified CME router, you need to configure **no service phone lineMode**.

Note The parameters for the **service phone** CLI command are case sensitive. For example, the command to configure ELM for Cisco IP Phone Series 8800 must be **service phone lineMode 1** . If the command input is **service phone LineMode 1**, **service phone linemode 1**, and so on, ELM is not configured.

Use the **show telephony-service tftp-binding** command to view the SEP*.cnf.xml files that are associated with individual phones. The following example entry from a Sep*.conf.xml file disables the PC port on a phone:

```
<vendorConfig>
<pcPort>1</pcPort>
</vendorConfig>
```

The below table lists the basic vendorConfig parameters in alphabetical order.

Note Parameter names are word and case-sensitive and must be typed exactly as shown.

Table 15: vendorConfig Parameter-Name and Parameter-Value Descriptions

Parameter Name and Value	Description
actionableAlert {**0** | 1}	Replaces the traditional incoming call pop-up notification with an alert that you must respond to. • **0**—Disabled. • **1**—Enabled (default).
adminPassword *password*	(For the Cisco Unified IP Phone 7921G only) Creates a password for accessing the web interface on a phone. • *passwor*d—String of up to 32 characters.
autoSelectLineEnable {**0** | 1}	Enables and disables auto line selection. • **0**—Disabled. • **1**—Enabled (default).
backlightIdleTimeout *HH*:*MM*	Sets the length of time in hours and minutes after which the backlighting of the IP phone displays will switch off again once the phone is inactive. • This parameter is applicable only on the days specified using the **daysBacklightNotActive** parameter. • This parameter does not affect the display during the period of time specified using the **backlightOnDuration** parameter. • Hour (*HH*) and minute (*MM*). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is one hour (01:00).
backlightOnDuration *HH*:*MM*	Sets the length of time in hours and minutes for which IP phone displays will be backlit. • This parameter does not affect the display on the days specified using the **daysBacklightNotActive** parameter. • Hour (*HH*) and minute (*MM*). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is 10 hours (10:00).
backlightOnTime *HH*:*MM*	Sets the time of day at which backlighting of the IP phone displays is switched on, using a 24-hour time format. • This parameter does not affect the display on the days specified using the **daysBacklightNotActive** parameter. • Hour (*HH*) and minute (*MM*). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is 07:30.
daysBacklightNotActive *number*[**,***number*...]	Sets the days of the week on which backlighting of the IP phone displays is switched off unless there is user interaction with the IP phone. • *number*—Represents the days of the week numerically, starting with Sunday (1) and ending with Saturday (7). Each number must be separated with a comma, without spaces: **daysBacklightNotActive 1,2,3**. • Default is no backlighting on Sun (1) and Sat (7).

Parameter Name and Value	Description
daysDisplayNotActive *number*[**,***number*...]	Sets the days of the week on which IP phone displays will be blank. • *number*—Represents the days of the week numerically, starting with Sunday (1) and ending with Saturday (7). Each number must be separated with a comma, without spaces: **daysDisplayNotActive 1,2,3** • Default is an inactive display on Sun (1) and Sat (7). • To disable this parameter so that IP phone displays are always active, configure this parameter in telephony-service configuration mode using a space plus a comma (,): **daysDisplayNotActive ,** for the *parameter-value*. **Note** This parameter is not supported in ephone-template configuration mode.
displayIdleTimeout *HH***:***MM*	Sets the length of time in hours and minutes for which IP phone displays will remain active, starting from the last time that the phone was used. • Hour (*HH*) and minute (*MM*). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is one hour (01:00). **Note** This parameter is not supported in ephone-template configuration mode.
displayOnDuration *HH***:***MM*	Sets the length of time in hours and minutes for which IP phone displays will be active. • Hour (*HH*) and minute (*MM*). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is 10 hours (10:00). **Note** This parameter is not supported in ephone-template configuration mode.
displayOnTime *HH***:***MM*	Sets the time of day at which IP phone displays are activated, using a 24-hour time format. • Hour (*HH*) and minute (*MM*). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05. Default is 07:30. **Note** This parameter is not supported in ephone-template configuration mode.
displayOnWhenIncomingCall {**0** | **1**}	Enables and disables an IP phone display to be activated when an incoming call is received (Line state is Ring in). The display will switch off again once the ringing stops if the user has not touched the phone and if the phone display is supposed to be off. • **0**—Disabled (default). • **1**—Enabled. **Note** This parameter is not supported in ephone-template configuration mode.
disableSpeaker {**true** | **false**}	Enables and disables the speakerphone. • **true**—Disabled. • **false**—Enabled (default).
disableSpeakerAndHeadset {**true** | **false**}	Enables and disables the speakerphone and headset. • **true**—Disabled. • **false**—Enabled (default).

Parameter Name and Value	Description
enableGroupListen {**true** | **false**}	(For Cisco Unified IP Phone 7906 and 7911 only) Enables and disables Group Listen mode in which the handset and speaker are both active to allow multiple listeners to hear the conversation over the speaker while one user talks on the handset. • **true**—Enabled. • **false**—Disabled (default). **Note** To support Group Listen, the speaker and headset must be enabled. See the **diasableSpeakerandHeadset** parameter for this command.
forwardingDelay {**0** | **1**}	Enables and disables the activation of the IP phone's PC Ethernet switch port when the IP phone boots to prevent Ethernet traffic from interfering with the bootup process. • **0**—Disabled. • **1**—Enabled (default).
garp {**0** | **1**}	Enables and disables IP phone response to gratuitous Address Resolution Protocol (ARP) messages from the IP phone's Ethernet interface. • **0**—Disabled. • **1**—Enabled (default).
g722CodecSupport {**0** | **1** | **2**}	Enables and disables the registration of the G.722 codec on the IP phone. • **0**—Phone default (default), equal to disabled or enabled and set by manufacturer. • **1**—Disabled. Disables G.722-64K2 codec on phone. • **2**—Enabled. Enables G.722-64K codec on phone.
handsetWidebandEnable {**0** | **1** | **2**}	Enables or disables wideband handset option on supported IP phones. • If the **handsetWidebandUIControl** parameter is set to Enable (1), the option set in the phone UI, by the phone user, has priority over the value set for this parameter. • **0**—Phone default (default), equal to disabled or enabled and set by manufacturer. • **1**—Enabled. Enables wideband handset on phone. • **2**—Disabled. Disables wideband headset on phone. • Wideband handset should only be used on supported IP phones with firmware version 8.3 or a later version.
handsetWidebandUIControl {**0** | **1**}	Enables or disables control of handset options by phone user. • **0**—Enabled (default). Allows phone user to select either narrowband or wideband handset in the phone UI. • **1**—Disabled.
headsetWidebandEnable {**0** | **1**}	Enables or disables wideband headset option on supported IP phones. • If the **headsetWidebandUIControl** parameter is set to Enable (0), the option set in the phone UI, by the phone user, has priority over the value set for this parameter. • **0**—Enabled (default). Enables wideband headset on phone. • **1**—Disabled. Disables wideband headset on phone. • Wideband handset should only be used on supported IP phones with firmware version 8.3 or a later version.

Parameter Name and Value	Description
headsetWidebandUIControl {**0** | **1**}	Enables or disables control of headset option by phone user. • **0**—Enabled (default). Allows phone user to select either narrowband or wideband headset • **1**—Disabled.
homeScreen {**0** | **1**}	(For Cisco Unified Wireless Phone 7921G only) Specifies view to be displayed on phone home screen. • **0**—Display main phone screen (default). • **1**—Display line view. • Implemented only on supported IP phones with firmware version 1.2.1 or a later version.
LineKeyBarge {**0** | **2**}	Activates the Line keys on the phones so that it displays the remote-in-use state softkeys correctly, and supports Barge functionality on Cisco IP Phone 7800 Series phones. The command is disabled by default. • **0**—Enables cBarge. • **2**—Enabales Barge. • **no service phone LineKeyBarge**—Disables Line Keys, so that the 7800 series IP phones will not display the remote-in-use state softkeys. . **Note** If the remote-in-use state softkey configuration has both Barge and cBarge configured, then cBarge is taken as the preferential feature. The phones will ignore the Barge configuration.
lineMode { **1**}	(For Cisco IP Phone 8800 Series only) Enables Enhanced Line Mode (ELM) for Unified CME. The no form of the command disables the ELM functionality. By default, ELM is disabled for Unified CME. • **no service phone lineMode**—Disables Enhanced Line Mode (default). • **service phone lineMode 1**—Enables Enhanced Line Mode.
loadServer [*hostname* | *IPaddress*]	(For the Cisco Unified IP Phone 7921G only) Directs the IP phone to use an alternative TFTP server such as a local server to obtain firmware loads and upgrades. Using this parameter can help to reduce installation time, particularly for upgrades over a WAN. The specified server must be running TFTP services and have the firmware file in the TFTP path. **Note** If the firmware file is not found, the firmware will not install. The phone will not be redirected to the TFTP server specified by the **option 150 ip** command. • *hostname*—Name of the server from which the IP phone must retrieve phone firmware. Maximum length: 256 characters. • *IPaddress*—IP address of server from which the IP phone must retrieve phone firmware. • To disable this command and redirect the phone to use the TFTP server specified by the **option 150 ip** command to obtain its load files and upgrades, use this parameter name without the *hostname* or *IPaddress* argument.

Parameter Name and Value	Description
pcPort {**0** | **1**}	Enables and disables the Ethernet switch port on the phone so the IP phone can have access to an Ethernet connection for a PC connection through the phone. • **0**—Enabled (default). • **1**—Disabled.
PushToTalkURL *url*	(For the Cisco Unified IP Phone 7921G only) Provisions the URL to be contacted for application services such as Push-To-Talk services. • *url*—URL as defined in RFC 2396. Maximum length is 256 characters.
settingsAccess {**0** | **1** | **2**}	Enables and disables the Settings button on an IP phone. • **0**—Disabled. • **1**—Enabled (default). The phone user can modify features by using the Settings menu. • **2**—Restricted. The phone user is allowed to access User Preferences and volume settings only.
spanToPCPort {**0** | **1**}	Enables and disables the path between the Ethernet switch port of an IP phone and a connection to a PC. • **0**—Enabled (default). • **1**—Disabled. **Note** The path must be disabled to support Desktop Monitoring and Recording in a Cisco UCCX/Cisco Unified CME integration.

specialNumbers *number*[**,***number*...]	(For the Cisco Unified IP Phone 7921G only) Identifies a number that can be dialed on a phone regardless of whether the phone is locked or unlocked. For example, in the United States, the 911 emergency number is a good special number candidate to be dialed without unlocking the phone.
	• *number*—Numerical string. Maximum length: 16 characters.
	• To identify more than one special number, separate the numbers with a comma (**,**). Do not include spaces between numbers.
	• The following example shows how to configure 411, 511, and 911 as special numbers:
	`Router(config)# telephony-service`
	`Router(config telephony-service)# service phone`
	` specialNumbers 411,511,911`

sshAccess {**0** | **1**}	Enables and disables SSH access. • **0**—Enabled (default). • **1**—Disabled.
thumbButton1 **PTTH***button_number*	(For Cisco Unified Wireless IP Phone 7921 and 7925) Associates thumb button on Cisco wireless IP phone with a phone button for one-way Push-To-Talk (PTT) functionality in Cisco Unified CME without requiring an external server. • *button_number*—Button on phone that is configured with an intercom dn that targets a paging number when user presses the thumb button. Range is 1 to 6. • The **PTTH***button_number* keyword/argument combination is a contiguous character string and cannot contain spaces. • Implemented on supported phones with firmware version 1.0.4 or a later version.

Parameter Name and Value	Description
videoCapability {**0** | **1**}	Enables and disables video capability for all applicable IP phones associated with a Cisco Unified CME router. • 0—Disabled (default). • 1—Enabled. • After using this parameter to enable video at a system level, you must configure the **video** command in ephone configuration mode for each video-capable phone. **Note** This parameter is not supported in ephone-template configuration mode.
voiceVlanAccess {**0** | **1**}	Enables and disables spanning, which is the IP phone's access to the voice VLAN of the PC to which the IP phone's Ethernet port is connected. • **0**—Enabled (default). • **1**—Disabled. **Note** For Cisco Unified IP Phone 7985, the default is Disabled (**1**).
webAccess {**0** | **1** | **2**}	Enables and disables web access that allows phone users to configure settings and features on User Option web pages. • **0**—Enabled (default). • **1**—Disabled. • **2**—Read Only. For the Cisco Unified IP Phone 7921G only. The phone user can view only User Option web pages and cannot modify settings and features on the pages. **Note** For the Cisco Unified IP Phone 7921G, the default is Read Only (**2**).
WLanProfile *tag* {**0** | **1**}	(For Cisco Unified IP Phone 7921G only) Locks or unlocks a specific profile. • *tag*—Unique number assigned to profile. Range is 1 to 4. • **0**—Locked (default). • **1**—Unlocked. User can modify a profile. • Repeat this command for each profile to be locked or unlocked.

Examples

The following example shows how to configure multiple **service phone** parameters. This configuration is applied only in as much as IP phone firmware supports each parameter.

```
Router(config)# telephony-service
Router(config-telephony)# service phone disableSpeaker true
Router(config-telephony)# service phone disableSpeakerAndHeadset true
Router(config-telephony)# service phone forwardingDelay 1
Router(config-telephony)# service phone garp 1
Router(config-telephony)# service phone pcPort 1
Router(config-telephony)# service phone voiceVlanAccess 0
Router(config-telephony)# service phone settingsAccess 1
Router(config-telephony)# service phone videoCapability 1
Router(config-telephony)# service phone daysDisplayNotActive 1,7
Router(config-telephony)# service phone displayOnTime 07:30
Router(config-telephony)# service phone displayOnDuration 10:00
Router(config-telephony)# service phone displayIdleTimeout 01:00
Router(config-telephony)# service phone daysBacklightNotActive 1,7
Router(config-telephony)# service phone backlightOnTime 07:30
```

```
Router(config-telephony)# service phone backlightOnDuration 10:00
Router(config-telephony)# service phone backlightIdleTimeout 01:00
Router(config-telephony)# create cnf-files
Router(config-telephony)# reset all
The following example shows how to set the default values for backlighting the phone display
 for all Cisco Unified IP phones with backlight capabilities in Cisco Unified CME:
Router(config)# telephony-service
Router(config-telephony)# service phone daysBacklightNotActive 1,7
Router(config-telephony)# service phone backlightOnTime 07:30
Router(config-telephony)# service phone backlightOnDuration 10:00
Router(config-telephony)# service phone backlightIdleTimeout 01:00
Router(config-telephony)# create cnf-files
Router(config-telephony)# reset all
```

The following example shows how to set the backlighting parameters so that there is no backlighting of the phone display for all Cisco Unified IP phones with backlight capabilities until there is user interaction with the phone. The **backlightIdleTimeout** parameter is configured so that the backlight will switch off again after 60 seconds of inactivity.

```
Router(config)# telephony-service
Router(config-telephony)# service phone daysBacklightNotActive 1,2,3,4,5,6,7
Router(config-telephony)# service phone backlightOnTime 07:30
Router(config-telephony)# service phone backlightOnDuration 10:00
Router(config-telephony)# service phone backlightIdleTimeout 00.01
Router(config-telephony)# create cnf-files
Router(config-telephony)# reset all
```

The following example shows how to set the display parameters so that the phone display for all Cisco Unified IP phones with luminous displays are blank on Sunday (1), Monday (2), and Saturday (7):

```
Router(config)# telephony-service
Router(config-telephony)# service phone daysDisplayNotActive 1,2,7
Router(config-telephony)# create cnf-files
Router(config-telephony)# reset all
```

The following example shows how to disable the PC port on an individual IP phone (ephone 15) using an ephone template:

```
Router(config)# ephone-template 8
Router(config-ephone-template)# service phone pcPort 1
Router(config-ephone-template)# exit
Router(config)# ephone 15
Router(config-ephone)# ephone-template 8
Router(config-ephone)# exit
Router(config)# telephony-service
Router(config-telephony)# create cnf-files
Router(config-telephony)# exit
Router(config)# ephone 15
Router(config-ephone)# reset
```

The following examples shows how to enable ELM on Unified CME for Cisco IP Phones. Also, it provides steps to configure **create profile** and **restart** the phones under **voice register global** configuration mode to enable ELM for the Cisco IP Phone 8800 series phones on Unified CME:

```
Router(config)#telephony-service
Router(config-telephony)#service phone lineMode ?
  WORD  enter the phone xml file parameter text for the previously entered
        parameter name
```

```
Router(config-telephony)#service phone lineMode 1
Router(config-telephony)#create cnf-files
Router(config-telephony)#end

Router(config)#voice register global
Router(config-register-global)#create profile
Router(config-register-global)#restart
Router(config-register-global)#end
```

Related Commands

Command	Description
cnf-file	Specifies that separate configuration files be generated for individual SCCP phones or types of SCCP phones.
create cnf-files	Builds XML configuration files that set IP phone displays and functionality.
create profile	Generates configuration profile files required for SIP phones
ephone-template (ephone)	Applies a template to the ephone being configured.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service tftp-binding	Displays the current configuration files accessible to IP phones.
show voice register tftp-bind	Displays the current configuration files accessible to IP phones.
video (ephone)	Enables video capabilities on specified phones.

service profile

To set the parameters under the commonProfile section in IP phone SEP*.cnf.xml configuration files, use the **service profile** command in telephony-service configuration mode. To disable the settings, use the **no** form of this command.

service profile [{**phonePassword** *password* | **callLogBlfEnabled** | **backgroundImageAccess** *false*}]
no service profile [{**phonePassword** *password* | **callLogBlfEnabled** | **backgroundImageAccess** *false*}]

Syntax Description

phonePassword *password*	Enters the phone password.
callLogBlfEnabled	Enables the call log.
backgroundImageAccess *false*	Disables the background image access.

Command Default

Parameters in the commonProfile section in IP phone SEP*.cnf.xml configuration files are not set.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Release	Modification
15.2(2)T2	This command was introduced.

Usage Guidelines

You can use the **service profile** command to set the parameters under the commonProfile section in IP phone SEP*.cnf.xml configuration files. Invoke the **create cnf-file** command to update phone configuration files.

Examples

The following example shows the **service profile** command is used at the router prompt:

```
Router# configure terminal
Router(config)# telephony-service
Router(config-telephony
)# service profile phonePassword cisco
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

service-digit

To enable phone users to dial a service digit to request off-net services, use the **service-digit** command in voice MLPP configuration mode. To reset to the default, use the **no** form of this command.

service-digit
no service-digit

Syntax Description

This command has no arguments or keywords.

Command Default

Service digit is disabled.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command enables users to request off-net services by dialing a service digit, after dialing the MLPP access digit. The service digit provides information to the switch when connecting calls to government or public telephone services or networks that are not part of the Defense Switched Network (DSN).

Phone users request a service by dialing the access code NS, where N is the preconfigured MLPP access digit and S is the service digit. The service digit is a number from 5 to 9.

In Cisco Unified CME, the dial plan must be configured to play secondary dial-tone and the rest of the dialed digits are collected and passed to the off-net trunk. The digits that follow the prefix NS must be E.164 compliant.

Examples

The following example shows how to enable users to dial a service digit:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# service-digit
```

Related Commands

Command	Description
access-digit	Defines the access digit that phone users dial to request a precedence call.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.

service-enable (auto-register)

To re-enable the auto-registration of SIP phones on Unified CME that is temporarily disabled, use the **service-enable** command in voice auto register configuration mode. This command is a sub-mode CLI of the command **auto-register** . To temporarily disable the auto registration process without losing configurations such as password and DN range, use the **no** form of this command.

service-enable
no service-enable

Syntax Description

no **service-enable**	Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.

Command Default

By default, this command is enabled.

Command Modes

voice auto register configuration (config-voice-auto-register)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

This command is enabled by default.

If the administrator needs to temporarily disable or enable auto registration without losing configurations such as DN range, and password, the no form of this command, *no* **service-enable** is used.

Examples

The following example shows how to temporarily disable auto registration using the no form of the sub-mode option, service-enable:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)# ?

VOICE auto register configuration commands:
auto-assign Define DN range for auto assignment
default Set a command to its defaults
exit Exit from voice register group configuration mode
no Negate a command or set its defaults
password Default password for auto-register phones
service-enable Enable SIP phone Auto-Registration
template Default template for auto-register phones

Router(config-voice-auto-register)#no service-enable ?
 <cr>
```

Related Commands

Command	Description
auto-register	Enables automatic registration of SIP phones with the Cisco Unified CME system.
password (auto-register)	Configures the mandatory password that administrator sets for auto registration of SIP phones on Unified CME.
auto-assign (auto-register)	Configures the mandatory range of directory numbers for phones auto registering on Unified CME.
template (auto-register)	Creates a basic configuration template that supports all the configurations available on the voice register template.
auto-reg-ephone	Enables automatic registration of ephones with the Cisco Unified CME system.

service-domain

To set the global MLPP domain type and number, use the **service-domain** command in voice MLPP configuration mode. To reset to the default, use the **no** form of this command.

service-domain {**drsn** | **dsn**} **identifier** *domain-number*
no service-domain

Syntax Description

drsn	Defense Red Switched Network (DRSN).
dsn	Defense Switched Network (DSN). This is the default value.
domain-number	Number to identify the global domain, in three-octet format. Range: 0x000000 to 0xFFFFFF. Default: 0.

Command Default

Domain type is **dsn**; domain number is 0.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command sets the global domain type and number in Cisco Unified CME. Use the **mlpp service-domain** command to assign registered phones to different service domains. Any phone not configured with a specific service domain uses this global domain for MLPP calls.

Examples

The following example shows the global domain set to DSN with identifier 0010:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# service-domain dsn identifier 0010
```

Related Commands

Command	Description
mlpp service-domain	Sets the service domain and maximum precedence (priority) level for MLPP calls.
preemption trunkgroup	Enables preemption capabilities on a trunk group.
service-domain (voice class)	Sets the service domain name in the MLPP voice class.

service-domain (voice class)

To set the service domain name in the MLPP voice class, use the **service-domain** command in voice class configuration mode. To reset to the default, use the **no** form of this command.

service-domain {**drsn** | **dsn**}
no service-domain

Syntax Description

drsn	Defense Red Switched Network (DRSN).
dsn	Defense Switched Network (DSN).

Command Default

Domain name is **dsn**.

Command Modes

Voice class configuration (config-class)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command sets the domain name that is used for off-net MLPP calls.

After using this command, assign the voice class to an outbound POTS or VoIP dial peer by using the **voice-class mlpp** command.

Examples

The following example shows the domain name set to DSN:

```
Router(config)# voice class mlpp
Router(config-class)# service-domain dsn
```

Related Commands

Command	Description
mlpp service-domain	Sets the domain number and maximum precedence (priority) level for an MLPP call.
service-domain	Sets the global MLPP domain type and number.
voice-class mlpp	Assigns an MLPP voice class to a POTS or VoIP dial peer.

service-domain midcall-mismatch

To define the behavior when there is a domain mismatch between the two legs of a call, use the **service-domain midcall-mismatch** command in voice MLPP configuration mode. To reset to the default, use the **no** form of this command.

service-domain midcall-mismatch {**method1** | **method2** | **method3** | **method4**}
no service-domain midcall-mismatch

Syntax Description

method1	Domain remains unchanged for each of the connections and the precedence level of the lower priority call changes to that of the higher priority call. This is the default value.
method2	Domain and precedence level of the lower priority call changes to that of the higher priority call.
method3	Domain remains unchanged for each of the connections and the precedence levels change to Routine for both calls.
method4	Domains change to that of the connection for which supplementary service was invoked (for example, transferee in case of transfer). Precedence levels change to Routine for both calls.

Command Default

The default is **method1**.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command determines the service domain and precedence level to apply in the case of a mismatch of these values between the two connections (call legs) of a call. This typically occurs when supplementary services such as Call Transfer or Conferencing are invoked during a call.

Examples

The following example shows the domain mismatch method set to 2:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# service-domain midcall-mismatch method2
```

Related Commands

Command	Description
mlpp service-domain	Sets the domain number and maximum precedence (priority) level for an MLPP call.
preemption trunkgroup	Enables preemption capabilities on a trunk group.
service-domain	Sets the default MLPP domain name and number.

session-server

To specify a session manager to manage and monitor Register and Subscribe messages during a feature-server session, use the **session-server** command in voice register dn configuration mode, voice register pool configuration mode, or ephone-dn configuration mode. To return to the default, use the **no** form of this command.

session-server*session-server-tag[, ...session-server-tag]*
no session-server *session-server-tag*

Syntax Description

session-server-tag	Unique identifier of previously configured session manager in Cisco Unified CME. Range: 1 to 8. When configured in voice register dn configuration mode or in ephone-dn configuration mode, this argument can contain up to eight session-server-tags, separated by commas (**,**).

Command Default

Session manager is not assigned.

Command Modes

Ephone-dn configuration (ephone-dn)
Voice register dn configuration (voice-register-dn)
Voice register pool configuration (voice-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW2	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Cisco Unified CME 4.2 and later versions provide a general interface for interoperating with external feature servers, such as the Cisco Unified CCX application on Cisco CRS, including call monitoring and device monitoring based on SIP presence and dialog event package. A session manager in Cisco Unified CME can manage and monitor Register and Subscribe messages.

Before configuring this command, a session manager must already configured in Cisco Unified CME by using the **voice register session-server** command.

Use the **session-server** command in voice register pool configuration mode to specify that Register and Subscribe messages for an external feature-server route point must contain a Cisco-referenceID field. Registration or subscription will be granted only for the specified route point. The route point for which Register and Subscribe messages are to be managed by this session manager must already be configured as a SIP endpoint in Cisco Unified CME. Typically, the configuration for the route point is provided from the feature server. If the configuration for the route point is deleted or must be modified, it can be reconfigured directly in Cisco Unified CME by using Cisco IOS commands. Each route point can be managed by only one session manager. Each session manager can manage multiple route points.

Use the **session-server** command in ephone-dn configuration mode or in voice register dn configuration mode to specify that Subscribe messages for a directory number must contain a Cisco-referenceID field. Registration or subscription will be granted only for the specified directory number. The directory number for which Subscribe messages are to be monitored by this session manager must already be configured in Cisco Unified CME. Each directory number can be monitored by up to eight session managers. Each session manager can subscribe for multiple directory numbers.

Examples

The following example shows the configuration for specifying that session manager 1 can control a route point (voice register pool) for an external feature server:

```
voice register pool 1
 session-server 1
```

The following example shows the configuration specifying which session managers can monitor Register and Subscribe messages to directory numbers associated with Cisco Unified CCX agent phones. Notice that several session managers (1, 3, 5, and 7) can subscribe for both directory numbers.

```
ephone-dn 1
 session-server 1,2,3,4,5,6,7,8
.
ephone-dn 2
 session-server 1,3,5,7
```

Related Commands

Command	Description
voice register session-server	Enters voice register session configuration mode for the purpose of configuring a session manager.

session-transport

To specify the transport layer protocol that a SIP phone uses to connect to Cisco Unified CME, use the **session-transport** command in voice register pool or voice register template configuration mode. To reset to the default value, use the **no** form of this command.

session-transport {**tcp** | **udp**}
no session-transport

Syntax Description

tcp	Transmission Control Protocol (TCP) is used.
udp	User Datagram Protocol (UDP) is used. This is the default.

Command Default

UDP is the default protocol.

Command Modes

Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command sets the transport layer protocol parameter in the phone's configuration file.

If you use a voice register template to apply a to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

This command is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

Note Although this command is not supported for the Cisco Unified IP Phone 7905, 7912, 7940, or 7960, it can be used to assign TCP as the session transport type for these phones. If TCP is selected for an unsupported phone using this command, calls to that phone will not complete successfully. The phone can originate calls but it uses UDP, although TCP has been assigned.

Examples

The following example sets the transport layer protocol to TCP for SIP phone 10:

```
Router(config)# voice register pool 10
Router(config-register-pool)# session-transport tcp
```

Related Commands

Command	Description
create profile	Generates the configuration profile files required for SIP phones.
show sip-ua status	Displays the status of SIP call service on a SIP gateway.
template (voice register pool)	Applies template to voice register pool being configured.

Cisco Unified CME Commands: S2

shared-line

To create a directory number to be shared by multiple SIP phones, use the **shared-line** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

shared-line [**max-calls** *number-of-calls*]
no shared-line

Syntax Description

max-calls *number-of-calls*	(Optional) Maximum number of active calls allowed on the shared line. Range: 2 to 16. Default: 2.

Command Default

Directory number is not a shared line. Maximum number of calls on a shared line is 2.

Command Modes

Voice register dn configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command enables a shared line on an individual SIP phone directory number.

This command is supported only on the Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE.

Examples

The following example shows that extension 5001 associated with directory number 2 is defined as a shared line and can support up to four calls:

```
Router(config)# voice register dn 2
Router(config-register-dn)# number 5001
Router(config-register-dn)# shared-line max-calls 4
```

Related Commands

Command	Description
busy-trigger-per-button	Sets the maximum number of calls allowed on a SIP shared line before activating Call Forward Busy or a busy tone.
debug shared-line	Displays debugging information about shared lines on SIP phones.
huntstop	Disables call hunting behavior for a directory number on a SIP phone.
number (voice register din)	Associates a telephone or extension number with a SIP phone.
show shared-line	Displays information about shared lines on SIP phones.
show voice register dn	Displays all configuration information associated with a specific voice register dn.

shared-line sip

To add an ephone-dn as a member of a shared directory number in the database of the Shared-Line Service Module for a mixed shared line between Cisco Unified SIP and Cisco Unified SCCP IP phones, use the **shared-line sip** command in ephone-dn configuration mode. To return to the default, use the **no** form of this command.

shared-line sip [**max calls** *number-of-calls*]
no shared-line sip

Syntax Description

max calls *number-of calls*	(Optional) Maximum number of active calls allowed on the shared line. Range: 2 to 16. Default: 2.

Command Default

Directory number is not a mixed shared line.

Maximum number of calls on a mixed shared line is 2.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **shared-line sip** command to add an ephone-dn as a member of a shared directory number in the database of the Shared-Line Service Module for a mixed shared line between Cisco Unified SIP IP phones and Cisco Unified SCCP IP phones. However, a mixed shared line is not enabled when an ephone-dn *nnnn* is the only shared directory number *nnnn* in the database of the Shared-Line Service Module. It is only enabled when a corresponding Cisco Unified SIP IP phone with a shared directory number *nnnn* is subscribed.

Mixed shared lines can only be configured on one of several common directory numbers. All attempts to add more are rejected.

Note The secondary number of an ephone-dn cannot be used as a search key in the Shared-Line Service Module.

Features are effectively supported on a mixed shared line when dial-plan patterns have matching configurations in telephony-service and voice register global configuration modes using the **dialplan pattern** command.

Examples

The following example shows 1001 as the shared line between a Cisco Unified SCCP IP phone and a Cisco Unified SIP IP phone. The maximum number of active calls allowed on the mixed shared line is four.

```
voice register dn 1
 number 1001
 shared-line max-calls 4
ephone-dn 1 octo-line
 number 1001
 shared-line sip
```

The following example shows how configuring a mixed shared line on a second common directory number is rejected:

```
Router(config)# ephone-dn 14 octo-line
Router(config-ephone-dn)# number 2502
Router(config-ephone-dn)# shared-line sip
Router(config)# ephone-dn 20 octo-line
Router(config-ephone-dn)# number 2502
Router(config-ephone-dn)# shared-line sip
DN number already exists in the shared line database
```

Related Commands

Command	Description
dialplan pattern	Defines a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, in telephony-service configuration mode.
dialplan pattern (voice register)	Defines a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, in voice register global configuration mode.
shared-line	Creates a directory number to be shared by multiple Cisco Unified SIP IP phones.

show capf-server

To display CAPF server configuration and session information, use the **show capf-server** command in privileged EXEC configuration mode.

show capf-server {**auth-string** | **sessions** | **summary**}

Syntax Description

auth-string	Display authentication strings for ephones.
sessions	Display information about active CAPF sessions.
summary	Display CAPF server configuration details.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example output displays CAPF server parameters:

```
Router# show capf-server summary

CAPF Server Configuration Details
        Trustpoint for TLS With Phone: cmeserver
        Trustpoint for CA operation: iosra
        Source Address: 10.1.1.1
        Listening Port: 3804
        Phone Key Size: 1024
        Phone KeyGen Retries: 100
        Phone KeyGen Timeout: 120 minutes
        Device Authentication Mode: Auth-String
```

The following example output displays the authentication strings that have been defined for the phones with the listed MAC addresses:

```
Router# show capf-server auth-string

Authentication Strings for configured Ephones
Mac-Addr        Auth-String
--------        -----------
000CCE3A817C    7012
001121116BDD    922
000D299D50DF    9182
000ED7B10DAC    3114
000F90485077    3328
0013C352E7F1    0678
```

The following example output displays active sessions between phones (identified by their MAC addresses) and the CAPF server. The phone ID field lists standard phone identifications, which include the letters “SEP” plus the MAC addresses of the phones. The below sample output defines the different session states that can appear in the output.

```
Router# show capf-server sessions

Active CAPF Sessions
Phone ID                State
SEP000CCE3A817C         AWAIT-KEYGEN-RES
```

Table 16: show capf-server sessions State Descriptions

State	Description
IDLE	Phone is idle.
AWAIT AUTH RES	A TLS connection was established on the TCP port that is specified in the configuration file. After a successful handshake verified the server certificate, a dialog was started between the CAPF server and the phone’s CAPF client. The server has challenged the phone by sending an authentication request and is waiting for a response.
AWAIT KEYGEN RESP	Phone authentication was successful. The CAPF server has sent a key generation request message to the phone and is waiting for a response.
AWAIT ENCRYPT MSG RESP	A key has been generated and the CAPF has used the phone’s public key to start the enrollment process with PKI. The CAPF sent an encrypt-message request to the phone and is waiting for a response.
AWAIT CA RESP	The phone has signed the received message using its private key and the CAPF has continued the enrollment process. PKI has forwarded the certificate request to the CA and is waiting for a response.
AWAIT STORE CERT RESP	Upon receiving an certificate issued from the CA, the CAPF has sent a store-certificate request message to the phone. The store-certificate request contains the certificate to be written to the phone’s flash memory. The CAPF is waiting for a store-certificate response message to confirm that the certificate has been stored.

show credentials

To display the credentials settings that have been configured for use during Cisco Unified CME phone authentication communications or secure Cisco Unified SRST fallback, use the **show credentials** command in privileged EXEC mode.

show credentials

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco SRST 3.3	This command was introduced for Cisco SRST.
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced for Cisco Unified CME.
12.4(9)T	Cisco Unified CME 4.0	This command for Cisco Unified CME was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Cisco Unified CME

This command displays the credentials settings on a Cisco Unified CME router that has been configured with a CTL provider to be used with Cisco Unified CME phone authentication.

Cisco Unified SRST

This command displays the credentials settings on the Cisco Unified SRST router that are supplied to Cisco Unified CallManager for use during secure SRST fallback.

Examples

The following is sample output from the **show credentials**:

```
Router# show credentials

Credentials IP: 10.1.1.22
Credentials PORT: 2445
Trustpoint: srstca
```

The below table describes the fields in the sample output.

Table 17: show credentials Field Descriptions

Field	Description
Credentials IP	Cisco Unified CME—IP address where the CTL provider is configured. Cisco Unified SRST—The specified IP address where certificates from Cisco Unified CallManager to the SRST router are received.

Field	Description
Credentials PORT	Cisco Unified CME—TCP port for credentials service communication. Default is 2444. Cisco Unified SRST—The port to which the SRST router connects to receive messages from the Cisco Unified IP phones. The port number is from 2000 to 9999. The default port number is 2445.
Trustpoint	Cisco Unified CME—CTL provider trustpoint label that will be used for TLS sessions with the CTL client. Cisco Unified SRST—The name of the trustpoint that is associated with the credentials service between the Cisco Unified CallManager client and the SRST router.

Related Commands

	Description
credentials	Enters credentials configuration mode to configure a Cisco Unified CME CTL provider certificate or a Cisco Unified SRST router certificate.
ctl-service admin	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
debug credentials	Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider and the CTL client or between a Cisco Unified SRST router and Cisco Unified CallManager.
ip source-address (credentials)	Enables the Cisco Unified CME or SRST router to receive messages through the specified IP address and port.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with a Cisco Unified SRST router certificate.

show cti

To display the status of the CTI subsystem, use the **show cti** command in privileged EXEC mode.

show cti {**call** | **gcid** | **line node** | **session**}

Syntax Description

call	Details for active (ACT) calls only.
gcid	List of Global Call IDs for active calls only.
line node	List of line nodes.
session	Details for active CTI sessions.

Command Modes

Privileged EXEC (#)

Command History

Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	This command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Usage Guidelines

This commands displays status information for the CTI subsystem in Cisco Unified CME.

Examples

The following sample output is for each command when there are no active calls.

```
Router#show cti gcid
GCID                                 callIDs
===================================  =================
no active GCID
Router#show cti call
DN         CallID GCID                                Calling    Called     State
=========  ====== =================================== ========== ========== =====
201
204
A line-node is the internal data structure of a directory number. Once a line-node is
created, the structure remains until the CTI interface is shut down.
Router#show cti line-node
line dn              number of call instance
================     =======================
1001                 0
201                  0
202                  0
203                  0
204                  0
233                  0
6789                 0
A0001                0
```

```
Router#
```

The following is sample output from the **show cti gcid** command for one call. This sample contains a single Gcid with two callIDs, one for each call leg.

```
Router#show cti gcid
GCID                                  callIDs
===================================  =================
1E2E3483-5ACB11DE-BA9EF925-DF2AFB55  59291, 59292,
```

The following is sample output from the **show cti call** command. This samples shows that a call was placed from (DN) 201 to (DN) 204 and both directory numbers are now Active (ACT). Note that the Gcid and callIDs in this sample correspond to those in the output from the **show cti gcid** command.

```
Router#show cti call
DN         CallID GCID                                Calling    Called     State
=========  ====== =================================== ========== ========== ======
201
           59291   1E2E3483-5ACB11DE-BA9EF925-DF2AFB55 201        204        ACT
204
           59292   1E2E3483-5ACB11DE-BA9EF925-DF2AFB55 201        204        ACT
```

The following is sample output from the **show cti line-node** command. In the following sample, there are eight line-nodes and two (201 and 204) are in use.

```
Router#show cti line-node
line dn              number of call instance
================     =======================
                     0
1001                 0
201                  1
      callID 59291(C7C ),  *cg = 201, cd = 204
202                  0
203                  0
204                  1
      callID 59292(C7C ),  cg = 201, *cd = 204
233                  0
6789                 0
A0001                0
```

<< Table number >> describes the significant fields shown in the display.

Table 18: show xxx Field Descriptions

Field	Description
GCID	Global Call ID (Gcid)—Unique identifier in for every call on an outbound leg of a VoIP dial peer for an endpoint. A single Gcid remains the same for the same call in the system, and is valid for redirect, transfer, and conference events.
CallID	Unique identifier for each call leg of a call.

Related Commands

Command	Description
clear cti session	Clears the session between a CSTA client application and Cisco Unified CME.

show ctl-client

To display information about the certificate trust list (CTL) client, use the **show ctl-client** command in privileged EXEC configuration mode.

show ctl-client

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example displays trustpoints and IP addresses known to the CTL client.

```
Router# show ctl-client

CTL Client Information
-----------------------------
        SAST 1 Certificate Trustpoint: cmeserver
        SAST 1 Certificate Trustpoint: sast2
        List of Trusted Servers in the CTL
                CME     10.1.1.1        cmeserver
                TFTP    10.1.1.1        cmeserver
                CAPF    10.1.1.1        cmeserver
```

show ephone

To display information about registered Cisco Unified IP phones, use the **show ephone** command in user EXEC or privileged EXEC mode.

show ephone [{*mac-addressphone-type*}]

Syntax Description

mac-address	(Optional) Displays information for the phone with the specified MAC address.
phone-type	(Optional) Displays information for phones of the specified phone type. Supported phone types are version-specific. Type **?** to display a list of values.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0 Cisco SRST 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0 Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(11)T	Cisco ITS 2.01 Cisco SRST 2.01	The **ata** keyword was added and this command was implemented on the Cisco 1760.
12.2(11)YT	Cisco ITS 2.1 Cisco SRST 2.1	The **7914** keyword was added.
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	The **7902**, **7905**, and **7912** keywords were added.
12.3(7)T	Cisco CME 3.1 Cisco SRST 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2.1 Cisco SRST 3.2.1	The **7970** keyword was added.
12.3(14)T	Cisco CME 3.3 Cisco SRST 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0 Cisco Unified SRST 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were added.
12.4(9)T	Cisco Unified CME 4.0 Cisco Unified SRST 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	The **7931** keyword was added for Cisco Unified CME.
12.4(4)XC4	Cisco Unified CME 4.0(3)	The **7931** keyword was added for Cisco Unified CME.
12.4(11)T	Cisco Unified CME 4.0(3)	The **7931** keyword for Cisco Unified CME was integrated into Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	The **7921** and **7985** keywords were added.

12.4(15)T1	Cisco Unified CME 4.1(1) Cisco Unified SRST 4.1(1)	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were added and this command was integrated into Cisco IOS Release 12.4(15)T1.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)	Emergency response location (ERL) information displays in the output.
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	Support for user-defined phone types created with the **ephone-type** command was added.
12.4(15)XZ1	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	The **7915-12**, **7915-24**, **7916-12**, **7916-24**, and **7937** keywords were added.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	The **7915-12**, **7915-24**, **7916-12**, **7916-24**, and **7937** keywords were added and this command was integrated into Cisco IOS Release 12.4(20)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. The IP-STE keyword was added and logical partitioning class of restriction (LPCOR) and Cancel Call Waiting information was added to the output.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Examples

Significant fields in the output from this command are described in the table.

The following sample output shows general information for registered phones:

```
Router# show ephone
ephone-8[7] Mac:000A.B7B1.444A TCP socket:[5] activeLine:0 whisperLine:0 REGISTERED in SCCP
 ver 11/9 max_streams=1
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0
paging 0 debug:0 caps:8 privacy:0
IP:10.4.188.99 * 50007 Telecaster 7940  keepalive 8424 max_line 2 available_line 2
button 1: cw:1 ccw:(0 0)
  dn 6  number 6006 CH1   IDLE         CH2   IDLE         overlay shared
button 2: cw:1 ccw:(0 0 0 0 0 0 0 0)
  dn 42 number 6042 CH1   IDLE         CH2   IDLE         CH3   IDLE         CH4   IDLE
      CH5   IDLE         CH6   IDLE         CH7   IDLE         CH8   IDLE         shared
overlay 1: 6(6006) 7(6007) 8(6008)
Preferred Codec: g711ulaw
Lpcor Type: local Incoming: ephone_group1 Outgoing: ephone_group1
```

The table describes significant fields in the output.

Table 19: show ephone Field Descriptions

Field	Description
Active Call	An active call is in progress.
activeLine	Line (button) on the phone that is in use. Zero indicates that no line is in use.
auto-dial *number*	Intercom extension that automatically dials *number*.

Field	Description
button *number*: dn *number*	Phone button number and the extension (ephone-dn) dn-tag number associated with that button.
bytes	Total number of voice data bytes sent or received by the phone.
Called Dn, Calling Dn	Ephone-dn tag numbers of the called and calling ephone-dn. Set to -1 if the call is not to or from an ephone-dn, or if there is no active call.
cfa *number*	Call-forward-all to *number* is enabled for this extension.
CH1 CH2	Status of channel 1 and, if this is a dual-line ephone-dn, the status of channel 2.
cw	1 indicates that Call Waiting is enabled. 0 indicates that Call Waiting is disabled.
debug	1 indicates that debug for the phone is enabled. 0 indicates that debug is disabled.
DnD	Do Not Disturb is set on this phone.
DP tag	Not used.
ephone-*number*	Unique sequence number used to identify this phone during configuration (phone-tag).
IP	Assigned IP address of the Cisco Unified IP phone.
Jitter	Amount of variation (in milliseconds) of the time interval between voice packets received by the Cisco Unified IP phone.
keepalive	Number of keepalive messages received from the Cisco Unified IP phone by the router.
Latency	Estimated playout delay for voice packets received by the Cisco Unified IP phone.
line *number*	Button number on an IP phone. Line 1 is the button nearest the top of the phone.
Lost	Number of voice packets lost, as calculated by the Cisco Unified IP phone, on the basis of examining voice packet time-stamp and sequence numbers during playout.
Lpcor Incoming	Setting of the **lpcor incoming** command.
Lpcor Outgoing	Setting of the **lpcor outgoing** command.
Lpcor Type	Setting of the **lpcor type** command.
Mac	MAC address.

Field	Description
Max Conferences	Maximum number of allowable conference calls and number of active conference calls.
max_line *number*	Maximum number of line buttons that can be configured on this phone.
mediaActive	1 indicates that an active conversation is in progress. 0 indicates that no conversation is ongoing.
monitor-ring	This button is set up as a monitor button.
number	Telephone or extension number associated with the Cisco Unified IP phone button and its dn-tag.
offhook	1 indicates that the phone is off-hook. 0 indicates that the phone is on-hook.
overlay	This button contains an overlay set. Use **show ephone overlay** to display the contents of overlay sets.
paging	1 indicates that the phone has received an audio page. 0 indicates that the phone has not received an audio page.
paging-dn	Ephone-dn that is dedicated for receiving audio pages on this phone. The paging-dn number is the number of the paging set to which this phone belongs.
Password	Authentication string that the phone user types when logging in to the web-based Cisco Unified CME GUI.
Port	Port used for TAPI transmissions.
REGISTERED	The Cisco Unified IP phone is active and registered. Alternative states are UNREGISTERED (indicating that the connection to the Cisco Unified IP phone was closed in a normal manner) and DECEASED (indicating that the connection to the Cisco Unified IP phone was closed because of a keepalive timeout).
reset	Pending reset.
reset_sent	Request for reset has been sent to the Cisco Unified IP phone.
ringing	1 indicates that the phone is ringing. 0 indicates that the phone is not ringing.
Rx Pkts	Number of received voice packets.
silent-ring	Silent ring has been set on this button and extension.
socket	TCP socket number used to connect to IP phone.
speed dial *speed-tag:digit-string label-text*	This button is a speed-dial button, assigned to the speed-dial sequence number *speed-tag*. It dials *digit-string* and displays the text *label-text* next to the button.

Field	Description
sub=3, sub=4	Subtype 3 means that one Cisco Unified IP Phone 7914 Expansion Module is attached to the main Cisco Unified IP Phones 7960 and 7960G, and subtype 4 means that two are attached.
Tag *number*	Dn-tag number, the unique sequence number that identifies an ephone-dn during configuration, followed by the type of ephone-dn it is.
TAPI Client IP Address	IP address of the PC running the TAPI client.
TCP socket	TCP socket number used to communicate with the Cisco Unified IP phone. This can be correlated with the output of other debug and show commands.
Telecaster *model-number*	Type and model of the Cisco Unified IP phone. This information is received from the phone during its registration with the router.
Tx Pkts	Number of transmitted voice packets.
Username	Username that the phone user types when logging in to the web-based Cisco Unified CME GUI.

Related Commands

Command	Description
show ephone-dn	Displays information about Cisco Unified IP phone extensions (ephone-dns).
show ephone login	Displays the login states of all local ephones.
show telephony-service all	Displays systemwide status and information for a Cisco Unified CME system.

show ephone attempted-registrations

To display the log of ephones that unsuccessfully attempt to register with Cisco Unified CME, use the **show ephone attempted-registrations** command in privileged EXEC mode.

show ephone attempted-registrations

Syntax Description

This command has no keywords or arguments.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **no auto-reg-ephone** blocks the automatic registration of ephones whose MAC addresses are not explicitly listed in the configuration. When automatic registration is blocked, Cisco Unified CME records the MAC addresses of phones that attempt to register but cannot because they are blocked.

Use the **show ephone attempted-registrations** to view the list of phones that have attempted to register but have been blocked. The **clear telephony-service ephone-attempted-registrations** clears the list.

Examples

The following example displays ephones that unsuccessfully attempted to register with Cisco Unified CME:

```
Router# show ephone attempted-registrations
Attempting Mac address:
Num    Mac Address         DateTime                         DeviceType
-----------------------------------------------------------------------------
1      C863.8475.5417      22:52:05 UTC Thu Apr 28 2005     SCCP Gateway (AN)
2      C863.8475.5408      22:52:05 UTC Thu Apr 28 2005     SCCP Gateway (AN)
.....
25     000D.28D7.7222      22:26:32 UTC Thu Apr 28 2005     Telecaster 7960
26     000D.BDB7.A9EA      22:25:59 UTC Thu Apr 28 2005     Telecaster 7960
...
47     C863.94A8.D40F      22:52:17 UTC Thu Apr 28 2005     SCCP Gateway (AN)
48     C863.94A8.D411      22:52:18 UTC Thu Apr 28 2005     SCCP Gateway (AN)
49     C863.94A8.D400      22:52:15 UTC Thu Apr 28 2005     SCCP Gateway (AN)
```

The below table describes the significant fields shown in the display.

Table 20: show ephone attempted-registrations Field Descriptions

Field	Description
Num	Index number.
Mac Address	MAC address of the ephone.
DateTime	Date and time that the attempt to register was made.

Field	Description
DeviceType	Type of ephone.

Related Commands

Command	Description
auto-reg-ephone	Enables automatic registration of ephones with the Cisco Unified CME system.
clear telephony-service ephone-attempted-registrations	Empties the log of ephones that unsuccessfully attempt to register with Cisco Unified CME.

show ephone cfa

To display status and information on the registered phones that have call-forward-all set on one or more of their extensions (ephone-dns), use the **show ephone cfa** command in privileged EXEC mode.

show ephone cfa

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following is sample output from the **show ephone cfa** command:

```
Router# show ephone cfa
ephone-1 Mac:0007.0EA6.353A TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:1.2.205.205 52491 Telecaster 7960  keepalive 14 max_line 6
button 1: dn 11 number 60011 cfa 60022 CH1 IDLE
button 2: dn 17 number 60017 cfa 60021 CH1 IDLE
```

The **show ephone** describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone dn

To display phone information for specified dn-tag or for all dn-tags, use the **show ephone dn** command in privileged EXEC mode.

show ephone dn [*dn-tag*]

Syntax Description

dn-tag	(Optional) Unique sequence number that is used during configuration to identify a particular extension (ephone-dn).

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Use this command to identify the phone on which a particular dn-tag has been assigned.

Examples

The following is sample output for the two appearances of DN 5:

```
Router# show ephone dn 5
Tag 5, Normal or Intercom dn
ephone 1, mac-address 0030.94C3.CAA2, line 2
ephone 2, mac-address 0030.94c2.9919, line 3
```

The **show ephone** describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone dnd

To display information on the registered phones that have “do not disturb” set on one or more of their extensions (ephone-dns), use the **show ephone dnd** command in privileged EXEC mode.

show ephone dnd

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command does not apply to Cisco Unified SRST.

Examples

The following is sample output from the **show ephone dnd** command:

```
Router# show ephone dnd
ephone-1 Mac:0007.0EA6.353A TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:1.2.205.205 52486 Telecaster 7960  keepalive 2729 max_line 6 DnD
button 1: dn 11 number 60011 CH1 IDLE
```

The **show ephone** describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone login

To display the login states of all local IP phones, use the **show ephone login** command in privileged EXEC mode.

show ephone login

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was modified. LOCAL and GLOBAL replace TRUE in the output for "Pin enabled."
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

The **show ephone login** command displays whether an ephone has a personal identification number (PIN) and whether its owner is logged in.

In Cisco Unified CME 7.1 and earlier versions, FALSE is displayed if there is no PIN configured for the specified ephone. TRUE is displayed if there is a PIN configured for the specified ephone.

In Cisco Unified CME 8.0 and later versions, the show output is modified as follows:

- FALSE is displayed only if no PIN is defined, neither in an ephone configuration nor in the telephony-service configuration.
- LOCAL is displayed if an individual PIN is defined for the specific ephone.
- GLOBAL is displayed if a global PIN is defined.

Cisco Unified CME 8.0 or Later Versions

The following is sample output from the **show ephone login** command. It shows that a PIN is defined for ephone 1 and that its owner has not logged in. The other phones do not have PINs associated with them.

```
Router# show ephone login
ephone 1        Pin enabled:LOCAL       Logged-in:FALSE
ephone 2        Pin enabled:FALSE
ephone 3        Pin enabled:FALSE
```

The following is sample output from the **show ephone login** command. It shows that a PIN is defined for ephone 1 and that its owner has not logged in. A global PIN is defined also defined for this system.

If the **pin** command is configured in ephone configuration mode and telephony-service configuration mode, the command in ephone configuration mode takes precedence.

```
Router# show ephone login
ephone 1        Pin enabled:LOCAL       Logged-in:FALSE
ephone 2        Pin enabled:GLOBAL      Logged-in:TRUE
ephone 3        Pin enabled:GLOBAL      Logged-in:TRUE
```

The following is sample output from the **show ephone login** command. It shows that neither a local nor a global PIN is enabled for ephones 1 to 3.

```
Router# show ephone login
ephone 1        Pin enabled:FALSE
ephone 2        Pin enabled:FALSE
ephone 3        Pin enabled:FALSE
```

Cisco CME 3.0 to Cisco Unified CME 7.1

The following is sample output from the **show ephone login** command. It shows that a PIN is enabled for ephone 1 and that its owner has not logged in. The other phones do not have PINs associated with them.

```
Router# show ephone login
ephone 1        Pin enabled:TRUE        Logged-in:FALSE
ephone 2        Pin enabled:FALSE
ephone 3        Pin enabled:FALSE
ephone 4        Pin enabled:FALSE
ephone 5        Pin enabled:FALSE
ephone 6        Pin enabled:FALSE
ephone 7        Pin enabled:FALSE
ephone 8        Pin enabled:FALSE
ephone 9        Pin enabled:FALSE
```

The below table describes significant fields in this output.

Table 21: show ephone login Field Descriptions

Field	Description
ephone *phone-tag*	Phone identified with its unique phone-tag sequence number.
Pin enabled	In Cisco Unified CME 7.1 and earlier versions: • TRUE—A PIN is defined for this phone. • FALSE —No PIN is defined for this phone. In Cisco Unified CME 8.0 and later versions: • LOCAL—A PIN has been defined for this phone. • GLOBAL—A global PIN is defined for this Cisco Unified CME system. • FALSE—No PIN is defined.

Field	Description
Logged-in	• TRUE indicates that a phone user is currently logged in on this phone. • FALSE indicates that no phone user is currently logged in on this phone.

Related Commands

Command	Description
login (telephony-service)	Defines when users of IP phones in a Cisco Unified CME system are logged out automatically.
pin	Sets set a personal identification number (PIN) for an IP phone in a Cisco Unified CME system.
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone moh

To display information about moh files in use, use the **show ephone moh** command in global configuration mode.

show ephone moh

Syntax Description

This command has no arguments or keywords

Command Modes

Global Configuration mode.

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use the show ephone moh to display information about the different MOH group configured. The following examples displays different MOH group configured.

Examples

```
Router #show ephone moh
Skinny Music On Hold Status (moh-group 1)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/minuet.au (not cached) type AU Media_Payload_G711Ulaw64k 160 bytes
Moh multicast 239.10.16.6 port 2000
Skinny Music On Hold Status (moh-group 2)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/audio/hello.au type AU Media_Payload_G711Ulaw64k  160 bytes
Moh multicast on 239.10.16.6 port 2000 via 0.0.0.0
Skinny Music On Hold Status (moh-group 3)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/bells.au type AU Media_Payload_G711Ulaw64k  160 bytes
Moh multicast on 239.10.16.5 port 2000 via 0.0.0.0
Skinny Music On Hold Status (moh-group 4)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/3003.au type AU Media_Payload_G711Ulaw64k  160 bytes
Moh multicast on 239.10.16.7 port 2000 via 0.0.0.0
Skinny Music On Hold Status (moh-group 5)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/4004.au type AU Media_Payload_G711Ulaw64k  160 bytes
Moh multicast on 239.10.16.8 port 2000 via 0.0.0.0
```

Related Commands

Command	Description
show ephone-dn	Displays MOH group information for a phone directory number.
show ephone summary	Displays the information about the MOH files in use
show voice moh-group statistics	Displays the MOH subsystem statistics information

show ephone offhook

To display information and packet counts for the phones that are currently off hook, use the **show ephone offhook** command in privileged EXEC mode.

show ephone offhook

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command is introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command is integrated into Cisco IOS Release 12.3(4)T.
Cisco IOS XE Gibraltar 16.11.1a	Unified CME 12.6	This command is enhanced to display the keys that are in use per media stream, along with the sRTP Ciphers.

Examples

The following sample output is displayed when no phone is off hook:

```
Router# show ephone offhook
No ephone in specified type/condition.
```

The following sample output displays information for a phone that is off hook:

```
Router# show ephone offhook
ephone-5 Mac:000A.8A2C.8C6E TCP socket:[20] activeLine:1 REGISTERED
mediaActive:0 offhook:1 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.22.84.71 51228 Telecaster 7960  keepalive 43218 max_line 6
button 1:dn 9  number 59943 CH1 SIEZE     silent-ring
button 2:dn 10 number 59943 CH1 IDLE
button 3:dn 42 number A4400  auto dial A4500 CH1 IDLE
button 4:dn 96 number 69943  auto dial 95259943 CH1 IDLE
button 5:dn 75 number 49943  auto dial 49943 CH1 IDLE
speed dial 1:57514 marketing
Active Call on DN 9 chan 1 :59943 0.0.0.0 0 to 0.0.0.0 2000 via 172.30.151.1
G711Ulaw64k  160 bytes vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1
Username:user1 Password:newuser
```

The following is a sample output for the show command, **show ephone offhook** . The lines that are added to the show command output as part of the Unified CME 12.6 enhancement are local key and remote key.

```
ephone-1[0] Mac:549A.EBB5.8000 TCP socket:[1] activeLine:1 whisperLine:0 REGISTERED in SCCP
 ver 21/17 max_streams=1 + Authentication + Encryption with TLS connection
mediaActive:1 whisper_mediaActive:0 startMedia:1 offhook:1 ringing:0 reset:0 reset_sent:0
paging 0 debug:0 caps:8
```

```
IP:8.44.22.63 * 17872 SCCP Gateway (AN) keepalive 28 max_line 1 available_line 1
port 0/0/0
button 1: cw:1 ccw:(0 0)
 dn 1 number 6901 CM Fallback CH1 CONNECTED CH2 IDLE
Preferred Codec: g711ulaw
Lpcor Type: none Active Secure Call on DN 1 chan 1 :6901 8.44.22.63 18116
 to 8.39.25.11 8066 via 8.39.0.1
G711Ulaw64k 160 bytes no vad
SRTP cipher: AES_CM_128_HMAC_SHA1_32
  local key: 0OPV0yxvcnRLPMzHfmYbwgHfdxcuS1uPbp5j/Tjk
  remote key: e8DQl3Kvk7LjZlipaCoMg9TMreBmiPsFmNiVHwIA
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1
```

The following sample output displays information for a phone that has just completed a call:

```
Router# show ephone offhook
ephone-5 Mac:000A.8A2C.8C6E TCP socket:[20] activeLine:1 REGISTERED
mediaActive:1 offhook:1 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.22.84.71 51228 Telecaster 7960  keepalive 43224 max_line 6
button 1:dn 9  number 59943 CH1 CONNECTED silent-ring
button 2:dn 10 number 59943 CH1 IDLE
button 3:dn 42 number A4400  auto dial A4500 CH1 IDLE
button 4:dn 96 number 69943  auto dial 95259943 CH1 IDLE
button 5:dn 75 number 49943  auto dial 49943 CH1 IDLE
speed dial 1:57514 marketing
Active Call on DN 9 chan 1 :59943 10.23.84.71 22926 to 172.30.131.129 2000 via 172.30.151.1
G711Ulaw64k  160 bytes no vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1 (media path callID 19288 srcCallID 1
9289)
Username:user1 Password:newuser
```

The **show ephone** describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone overlay

To display information for the registered phones that have overlay ephone-dns associated with them, use the **show ephone overlay** in privileged EXEC mode.

show ephone overlay

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command does not apply to Cisco Unified SRST.

Examples

The following is sample output from the show ephone overlay command:

```
Router# show ephone overlay
ephone-1 Mac:0007.0EA6.353A TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.2.225.205 52486 Telecaster 7960  keepalive 2771 max_line 6
button 1: dn 11 number 60011 CH1 IDLE      overlay
button 2: dn 17 number 60017 CH1 IDLE      overlay
button 3: dn 24 number 60024 CH1 IDLE      overlay
button 4: dn 30 number 60030 CH1 IDLE      overlay
button 5: dn 36 number 60036 CH1 IDLE      CH2 IDLE      overlay
button 6: dn 39 number 60039 CH1 IDLE      CH2 IDLE      overlay
overlay 1: 11(60011) 12(60012) 13(60013) 14(60014) 15(60015) 16(60016)
overlay 2: 17(60017) 18(60018) 19(60019) 20(60020) 21(60021) 22(60022)
overlay 3: 23(60023) 24(60024) 25(60025) 26(60026) 27(60027) 28(60028)
overlay 4: 29(60029) 30(60030) 31(60031) 32(60032) 33(60033) 34(60034)
overlay 5: 35(60035) 36(60036) 37(60037)
overlay 6: 38(60038) 39(60039) 40(60040)
```

The **show ephone** command describes significant fields in this output. The below table describes a field that is not in that table.

Table 22: show ephone overlay Field Descriptions

Field	Description
overlay *number*	Displays the contents of an overlay set, including each dn-tag and its associated extension number.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone phone-load

To display information about the phone firmware that is loaded on registered phones, use the **show ephone phone-load** command in privileged EXEC mode.

show ephone phone-load

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following is sample output that displays the phone firmware versions for all phones in the system:

```
Router# show ephone phone-load
DeviceName        CurrentPhoneload       PreviousPhoneload      LastReset
=====================================================================
SEP0002B9AFC49F   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP003094C2D0B0   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP000C30F03707   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP003094C2999F   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP000A8A2C8C6E   3.2(2.14)              3.2(2.14)              Initialized
SEP0002B9AFBB4D   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP00075078627F   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP0002FD659E59   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP00024BCCD626   3.2(2.14)                                     CM-closed-TCP
SEP0008215F88C1   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP000C30F0390C   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP003094C30143   3.2(2.14)              3.2(2.14)              TCP-timeout
```

The below table describes significant fields in this output.

Table 23: show ephone phone-load Field Descriptions

Field	Description
DeviceName	Device name.
CurrentPhoneLoad	Current phone firmware version.
PreviousPhoneLoad	Phone firmware version before last phone load.
LastReset	Reason for last reset of phone.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone registered

To display the status of registered phones, use the **show ephone registered** command in privileged EXEC mode.

show ephone registered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. The output was enhanced to include the setting of the feature-button command.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Examples

The following is sample output from the show ephone registered command:

```
Router# show ephone registered
ephone-12[11] Mac:001A.A11B.7D6D TCP socket:[5] activeLine:0 whisperLine:0 REGIS
TERED in SCCP ver 15/12 max_streams=1
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 res
et_sent:0 paging 0 debug:0 caps:7
IP:10.10.1.17 * 35177 6941  keepalive 3593 max_line 4 available_line 3
button 1: cw:1 dn 11 number 1001 CH1   IDLE         CH2   IDLE
button 2: cw:1 dn 56 number 6971  auto dial 6970 CH1   IDLE
button 3: cw:1 dn 10 number 1000 CH1   IDLE         CH2   IDLE
1 feature buttons enabled: dnd
Preferred Codec: g711ulaw
Lpcor Type: none
```

The below table describes significant fields in this output.

Table 24: show ephone registered Field Descriptions

Field	Description
active	Number of active parties registered.
ephone	Cisco IP phone.
mac-address	MAC address of the Cisco IP phone.
keepalive	Defines keepalive timeout period to unregister IP phone.

Field	Description
feature-buttons	Displays the type of feature button on the ephone.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone registered summary

To display the details of all the registered Skinny Client Control Protocol (SCCP) phones that are sorted based on ephone tags, use the **show ephone registered summary** command in privileged EXEC mode.

show ephone registered summary

Syntax Description

This command has no arguments or keywords.

Command Default

This command has no default behavior or values.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

Use this command to view the details of the registered phones configured in the SCCP mode sorted by ephone tags.

Example

The following is sample output of the registered phones configured in the SCCP mode.

Note The ***** symbol adjacent to the Directory Number (DN) in the command output indicates that the Directory Number (DN) is an Overlay-dn.

```
router# show ephone registered summary

==================================================================================
PhoneType  Ephone  MacAddress      IpAddress   Ln    Dn   Number   Status
==================================================================================
8941         1     7081.050C.0927  9.51.0.71    1    1    3001    Registered
                                                2    2*   3002    Registered
                                                2    5*   3005    Registered
                                                2    6*   3006    Registered
7970         2     001B.D52C.DF27  9.51.0.72    1    3    3003    Registered
                                                2    4    3004    Registered
7970         5     001B.D52C.4AEE  9.51.0.75    1    9    3009    Registered
                                                2    10   3010    Registered
==================================================================================
Total ephones configured    : 10
Total ephones registered    : 3
Total ephones unregistered  : 5
Total ephones deceased      : 0
Ephones in unknown state    : 2
```

Table 25: show ephone registered summary field descriptions

Field	Description
DN	Directory number of the phone.
Ephone	Total number of ephone tags configured.
IP Address	IP address of the phones.
LN	Line number of the phone.
MacAddress	Shows the MAC address of the SCCP phone.
Number	Number assigned to ephone.
PhoneType	Shows the type of Cisco IP phone.
Status	Shows the registration status.

Related Commands

Command	Description
show ephone summary types	Displays the total number of registered and unregistered SCCP phones for each phone type.
show ephone unregistered summary	Displays the details of all the unregistered SCCP phones.

show ephone remote

To display nonlocal phones (phones with no Address Resolution Protocol [ARP] entry), use the **show ephone remote** command in privileged EXEC mode.

show ephone remote

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Phones without ARP entries are suspected not to be on the LAN. Use the **show ephone remote** command to identify phones without ARP entries that might have operational issues.

Examples

The following is sample output that identifies ephone 2 as not having an ARP entry:

```
Router# show ephone remote
ephone-2 Mac:0185.047C.993E TCP socket:[4] activeLine:0 REGISTERED
mediaActive:1 offhook:0 ringing:0 reset:0 reset_sent:0 paging 1 debug:0
IP:10.50.50.20 49231 Telecaster 7910  keepalive 112 max_line 2 dual-line
button 1:dn 3  number 95021 CH1 IDLE
paging-dn 25
```

The **show ephone** describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone ringing

To display information on phones that are ringing, use the **show ephone ringing** command in privileged EXEC mode.

show ephone ringing

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following is sample output from the **show ephone ringing** command:

```
Router# show ephone ringing
ephone-1 Mac:0005.5E37.8090 TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:1 reset:0 reset_sent:0 paging 0 debug:0
IP:10.50.50.10 49329 Telecaster 7960  keepalive 17602 max_line 6
button 1:dn 1  number 95011 CH1 RINGING   CH2 IDLE
button 2:dn 2  number 95012 CH1 IDLE
```

The **show ephone** describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone rtp connections

To display active Real-Time Transport Protocol (RTP) call information on ephone call legs, use the **show ephone rtp connections** command in privileged EXEC mode.

show ephone rtp connections

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

The **show ephone rtp connections** command displays information on active RTP calls, including the ephone tag number of the phone with an active call, the channel of the ephone-dn, and the caller and called party's numbers for the connection for both local and remote endpoints. The output from this command provides an overview of all the connections in the system, narrowing the criteria for debugging pulse code modulation and Cisco Unified CME packets without a sniffer.

Note

When an ephone to non-ephone call is made, information on the non-ephone does not appear in a **show ephone rtp connections** command output. To display the non-ephone call information, use the **show voip rtp connections** command.

Examples

The following sample output shows all the connected ephones in the Cisco Unified CME system. The sample output shows five active ephone connections with one of the phones having the **dspfarm-assist** keyword configured to transcode the code on the local leg to the indicated codec. The output also shows four ephone to ephone calls, represented in the CallID columns of both the RTP connection source and RTP connection destination by zero values.

Normally, a phone can have only one active connection but in the presence of a whisper intercom call, a phone can have two. In the sample output, ephone-40 has two active calls: it is receiving both a normal call and a whisper intercom call. The whisper intercom call is being sent by ephone-6, which has an invalid LocalIP of 0.0.0.0. The invalid LocalIP indicates that it does not receive RTP audio because it only has a one-way voice connection to the whisper intercom call recipient.

```
Router# show ephone rtp connections
Ephone RTP active connections :
Ephone    Line  DN Chan  SrcCallID  DstCallID            Codec (xcoded?)
    SrcNum DstNum  LocalIP               RemoteIP
ephone-5     1   5    1         15         14              G729 (Y)
    1005  1102  [192.168.1.100]:23192  [192.168.1.1]:2000
ephone-6     2  35    1          0          0       G711Ulaw64k (N)
    1035  1036  [0.0.0.0]:0  [192.168.1.81]:21256
ephone-40    1 140    1          0          0       G711Ulaw64k (N)
    1140  1141  [192.168.1.81]:21244  [192.168.1.70]:20664
ephone-40    2  36    1          0          0       G711Ulaw64k (N)
    1035  1036  [192.168.1.81]:21256  [192.168.1.1]:2000
```

```
ephone-41    1 141    1          0          0       G711Ulaw64k (N)
    1140  1141  [192.168.1.70]:20664  [192.168.1.81]:21244
Found 5 active ephone RTP connections
```

The below table explains the fields in the **show ephone rtp connections** command output.

Table 26: show ephone rtp connections Field Descriptions

Field	Description
Ephone	Ephone tag number with an active call.
Line	Line appearance of the phone.
DN	Ephone-dn tag.
Chan	Channel of the ephone-dn.
SrcCallID	CCAPI CallID for the RTP connection source. For ephone to ephone calls, this will be 0. SrcCallID compares to “CallId” in the **show voip rtp connections** command output.
DstCallID	CCAPI CallID for the RTP connection destination. For ephone to ephone calls, this will be 0. DstCallID compares to “dstCallId” in the **show voip rtp connections** command output.
Codec (xcoded)	Codec name used by the phone with the active call. If xcoded is ‘Y’, the phone has the **dspfarm-assist** keyword configured to transcode the code on the local leg to the indicated codec.
SrcNum	Caller’s number for the connection. This number is not necessarily the ephone’s DN.
DstNum	Called party’s number for the connection.
LocalIP	Call’s local IP address and port. This is usually the ephone’s IP address. The IP address in brackets is either in IPv4 or IPv6 format, followed by a colon and the port number. The port compares to the “LocalRTP” number in the **show voip rtp connections** command output.
RemoteIP	Call’s remote IP address and port. For flow-around ephone to ephone calls, this is usually the other ephone’s IP address. For flow-through trunk calls, this is usually the Cisco Unified CME’s IP address. The port compares to the “RmtRTP” number in the **show voip rtp connections** command output.

Related Commands

Command	Description
show ephone registered	Displays the status of registered SCCP phones in Cisco Unified CME.
show voip rtp connections	Displays information about Real-Time Transport Protocol (RTP) named event packets.

show ephone socket

To display IP addresses (IPv4, IPv6, or dual-stack) being used by ephone sockets, use the **show ephone socket** command in privileged EXEC mode.

show ephone socket

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use the **show ephone socket** command to verify if IPv4 only, IPv6 only, or dual-stack (IPv4/IPv6) is configured on Cisco Unified CME. In the following example, skinny_tcp_listen_socket fd = 0 and skinny_tcp_listen_socket fd = 1 verify that dual-stack configuration. When IPv6 only is configured show ephone socket command displays skinny_tcp_listen_socket fd = -1 and skinny_tcp_listen_socket fd = 0 values. When IPv4 only is configured the show ephone socket command displays skinny_tcp_listen_socket fd = 0 and skinny_tcp_listen_socket (ipv6) fd = -1 values.

Examples

The following is sample output from the **show ephone socket** command:

```
Router# show ephone ssocket
skinny_tcp_listen_socket fd = 0
skinny_tcp_listen_socket (ipv6) fd = 1

skinny_secure_tcp_listen_socket fd = -1
skinny_secure_tcp_listen_socket (ipv6) fd = -1

skinny_open_sockets = 3:
Phone 3,
skinny_sockets[0] fd = 1
        read_buffer 0x480061E8, read_offset 0, read_header N, read_length 0

        resend_queue 0x47CE8178, resend_offset 0, resend_flag N, resend_Q_depth 0
Phone 2,
skinny_sockets[1] fd = 2
        read_buffer 0x48006A24, read_offset 0, read_header N, read_length 0

        resend_queue 0x47CE8104, resend_offset 0, resend_flag N, resend_Q_depth 0
Phone 1,
skinny_sockets[2] fd = 3
        read_buffer 0x48007260, read_offset 0, read_header N, read_length 0

        resend_queue 0x47CE8090, resend_offset 0, resend_flag N, resend_Q_depth 0
```

Related Commands

Command	Description
show ephone summary	Displays information about Cisco IP phones.

show ephone summary brief

To display details of all the SCCP phones sorted by ephone-tag, use the **show ephone summary brief** command in privileged EXEC mode.

show ephone summary brief

Syntax Description This command has no arguments or keywords.

Command Default This command had no default behavior or values.

Command Modes Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco SIP CME 10.5	This command was introduced.

Usage Guidelines The command output displays the status, IP address, and MAC address of the phones.

Example

The following is sample output of the **show ephone summary brief** command.

Note The asterisk symbol (*) adjacent to the Directory Number (DN) in the command output indicates that the Directory Number (DN) is an Overlay-dn.

```
router# show ephone summary brief
==========================================================================================
PhoneType         Ephone         MacAddress         IpAddress         Ln       Dn        Number
   Status
==========================================================================================
8941                 1        7081.050C.0927        9.51.0.71          1        1          3001
   Registered
                                                                       2        2*         3002
   Registered
                                                                       2        5*         3005
   Registered
                                                                       2        6*         3006
   Registered
7970                 2        001B.D52C.DF27        9.51.0.72          1        3          3003
   Registered
                                                                       2        4          3004
   Registered
7970                 3        001B.54CA.43F7                           1        5          3005
   Unregistered
                                                                       2        6          3006
   Unregistered
                                                                       3        2*         3002
   Unregistered
                                                                       3        3*         3003
   Unregistered
                                                                       3        8*         3008
```

```
   Unregistered
8945              4       D48C.B5C9.D2E6                         1       7        3007
   Unregistered
                                                                 2       8        3008

   Unregistered
7970              5       001B.D52C.4AEE        9.51.0.75        1       9        3009
   Registered
                                                                 2       10       3010

   Registered
8941              6       1111.2222.3333
   Unregistered
8941              10      1111.2222.3334
   Unregistered
6901                      11 1111.2222.3332
   Unregistered
Unknown Ephone    12
   Unknown
Unknown Ephone    13
   Unknown
==========================================================================================
Total ephones configured  : 10
Total ephones registered  : 3
Total ephones unregistered: 5
Total ephones deceased    : 0
Ephones in unknown state  : 2
```

Table 27: show ephone summary brief field descriptions

Field	Description
DN	Directory number of the phone.
Ephone	ephone tag.
IP Address	IP address of the phone.
LN	Line number of the phone.
MacAddress	Shows the MAC address of the SCCP phone.
Number	Number assigned to ephone.
PhoneType	Shows the type of Cisco IP phone.
Status	Shows the registration status.

Related Commands

Command	Description
show ephone summary types	Displays the total number of registered and unregistered SCCP phones for each phone type.

show ephone summary

To display brief information about Cisco IP phones, use the **show ephone summary** command in privileged EXEC mode.

show ephone summary

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0 Cisco SRST 1.0	This command was introduced.
12.2(8)T	Cisco CME 2.0 Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T .
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was modified. The output was enhanced to show IPv6 or IPv4 addresses configured on ephones.
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was modified. The output was enhanced to show voice-class stun-usage information.

Examples

The following is sample output from the **show ephone summary** command:

```
Router# show ephone summary
hairpin_block:
ephone-1[0] Mac:FCAC.3BAE.0000 TCP socket:[17] activeLine:0 whisperLine:0 REGISTERED
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0
debug:0  primary_dn: 1*
IP:10.2.1.0 * SCCP Gateway (AN)  keepalive 2966   music 0  1:1
port 0/0/0
voice-class stun is enabled
ephone-2[1] Mac:FCAC.3BAE.0001 TCP socket:[18] activeLine:0 whisperLine:0 REGISTERED
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0
debug:0  primary_dn: 2*
IP:10.2.1.5 * SCCP Gateway (AN)  keepalive 2966   music 0  1:2
port 0/0/1
voice-class stun is enabled
ephone-4 Mac:0030.94C3.F43A TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.2.1.1 Telecaster 7960  keepalive 59
Max 48, Registered 1, Unregistered 0, Deceased 0, Sockets 1
Max Conferences 4 with 0 active (4 allowed)
Skinny Music On Hold Status
Active MOH clients 0 (max 72), Media Clients 0
No MOH file loaded
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone summary types

To display the total count of registered and unregistered phones for each phone type operating in the Skinny Client Control Protocol (SCCP) mode, use the **show ephone summary types** command in privileged EXEC mode.

show ephone summary types

Syntax Description

This command has no arguments or keywords.

Command Default

This command has no default behavior or values.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

This command displays the count of configured, registered, unregistered, and deceased phones.

Example

The following is an example of the **show ephone summary types** command:

```
Router# show ephone summary types
===========================================================================
PhoneType            Configured  Registered  Unregistered  Deceased  Other
===========================================================================
Unknown Ephone type      2           0            0            0        2
6901                     1           0            1            0        0
8945                     1           0            1            0        0
7970                     3           2            1            0        0
8941                     3           1            2            0        0
===========================================================================
Total Phones             10          3            5            0        2
===========================================================================
```

Related Commands

Command	Description
show ephone summary brief	Displays the details of all the SCCP phones configured.

show ephone tapiclients

To display status of ephone Telephony Application Programming Interface (TAPI) clients, use the **show ephone tapiclients** command in privileged EXEC mode.

show ephone tapiclients

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following is sample output from the **show ephone tapiclients** command:

```
Router# show ephone tapiclients
ephone-4 Mac:0007.0EA6.39F8 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:192.168.1.18 50291 Telecaster 7960  sub=3 keepalive 728 max_line 20
button 1:dn 6  number 1004 CH1 IDLE      CH2 IDLE
button 2:dn 1  number 1000 CH1 IDLE      shared
button 3:dn 2  number 1000 CH1 IDLE      shared
button 7:dn 3  number 1001 CH1 IDLE      CH2 IDLE      monitor-ring shared
button 8:dn 4  number 1002 CH1 IDLE      CH2 IDLE      monitor-ring shared
button 9:dn 5  number 1003 CH1 IDLE      CH2 IDLE      monitor-ring
button 10:dn 91 number A00  auto dial A01 CH1 IDLE
speed dial 1:2000 PAGE-STAFF
speed dial 2:2001 HUNT-STAFF
paging-dn 90
Username:userB Password:ge30qe
Tapi client information
Username:userB status:REGISTERED Socket :[5]
 Tapi Client IP address: 192.168.1.5  Port:2295
```

The **show ephone** command describes significant fields in this output.

Related Commands

	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone telephone-number

To display information for the phone associated with a specified number, use the **show ephone telephone-number** command in privileged EXEC mode.

show ephone telephone-number *number*

Syntax Description

number	Telephone number that is associated with an ephone.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Use this command to find the phone on which a particular telephone number appears.

Examples

The following is sample output from the **show ephone telephone-number**:

```
Router# show ephone telephone-number 91400
DP tag: 0, primary
Tag 1, Normal or Intercom dn
  ephone 1, mac-address 000A.0E51.19F0, line 1
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone unregistered

To display information about unregistered phones, use the **show ephone unregistered** command in privileged EXEC mode.

show ephone unregistered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

There are two ways that an ephone can become unregistered. The first way is when an ephone is listed in the running configuration but no physical device has been registered for that ephone. The second way is when an unknown device was registered at some time after the last router reboot but has since unregistered.

Examples

The following is sample output from the **show ephone unregistered**:

```
Router# show ephone unregistered
ephone-1 Mac:0007.0E81.10F0 TCP socket:[-1] activeLine:0 UNREGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:0.0.0.0 0 Unknown 0  keepalive 0 max_line 0
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone unregistered summary

To display the details of all the unregistered Skinny Call Control Protocol (SCCP) phones sorted by ephone tag, use the **show ephone unregistered summary** command in privileged EXEC mode.

show ephone unregistered summary

Syntax Description

This command has no arguments or keywords.

Command Default

This command has no default behavior or values.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco SIP CME 10.5	This command was introduced.

Usage Guidelines

Use this command to view the details of the unregistered phones configured in the SCCP mode.

Example

The following is a sample output of the **show ephone unregistered summary** command.

Note

The ***** symbol adjacent to the Directory Number (DN) in the command output indicates that the Directory Number (DN) is an Overlay-dn.

```
router# show ephone unregistered summary
===============================================================================
PhoneType    Ephone    MacAddress     IpAddress    Ln    Dn    Number    Status
===============================================================================
7970          3       001B.54CA.43F7              1     5    3005     Unregistered
                                                  2     6    3006     Unregistered
                                                  3     2*   3002     Unregistered
                                                  3     3*   3003     Unregistered
                                                  3     8*   3008     Unregistered
8945          4       D48C.B5C9.D2E6              1     7    3007     Unregistered
                                                  2     8    3008     Unregistered
8941          6       1111.2222.3333                                  Unregistered
8941          10      1111.2222.3334                                  Unregistered
6901                  11 1111.2222.3332                               Unregistered
===============================================================================
Total ephones configured  : 10
Total ephones registered  : 3
Total ephones unregistered: 5
Total ephones deceased    : 0
Ephones in unknown state  : 2
```

Table 28: show ephone unregistered summary field descriptions

Field	Description
DN	Directory number of the phone.
Ephone	Total number of ephone tags configured.
IP Address	IP address of the phones.
LN	Line number of the phone.
MacAddress	Shows the MAC address of the SCCP phone.
Number	Number assigned to ephone.
PhoneType	Shows the type of Cisco IP phone.
Status	Shows the registration status.

Related Commands

Command	Description
show ephone registered summary	Displays the details of all the registered SCCP phones.
show ephone summary types pattern	Displays the total number of registered and unregistered SCCP phones for each phone type.

show ephone-dn

To display status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified Survivable Remote Site Telephony (SRST) environment, use the **show ephone-dn** command in privileged EXEC mode.

show ephone-dn [*dn-tag*]

Syntax Description

dn-tag	(Optional) For Cisco Unified CME, a unique sequence number that is used during configuration to identify a particular extension (ephone-dn). (Optional) For Cisco Unified SRST, a destination number tag. The destination number can be from 1 to 288.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0 Cisco SRST 1.0	This command was introduced.
12.2(8)T	Cisco CME 2.0 Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T command.

Examples

Cisco Unified CME

The following Cisco Unified CME sample output displays status and information for all ephone-dns:

```
Router# show ephone-dn
50/0/1 CH1 DOWN
EFXS 50/0/1 Slot is 50, Sub-unit is 0, Port is 1
 Type of VoicePort is EFXS
 Operation State is UP
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 8 ms
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
```

```
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US
 Station name None, Station number 91400
 Caller ID Info Follows:
 Standard BELLCORE
 Translation profile (Incoming):
 Translation profile (Outgoing):
 Digit Duration Timing is set to 100 ms
50/0/2 CH1 IDLE      CH2 IDLE
EFXS 50/0/2 Slot is 50, Sub-unit is 0, Port is 2
 Type of VoicePort is EFXS
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 8 ms
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US
 Station name None, Station number 91450
 Caller ID Info Follows:
 Standard BELLCORE
 Translation profile (Incoming):
 Translation profile (Outgoing):
 Digit Duration Timing is set to 100 ms
```

Cisco Unified SRST

The following SRST sample output displays status and information for all ephone-dns:

```
Router# show ephone-dn 7
50/0/7 INVALID
EFXS 50/0/7 Slot is 50, Sub-unit is 0, Port is 7
 Type of VoicePort is EFXS
```

```
Operation State is UP
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to 8 ms
Playout-delay Mode is set to default
Playout-delay Nominal is set to 60 ms
Playout-delay Maximum is set to 200 ms
Playout-delay Minimum mode is set to default, value 4 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Ringing Time Out is set to 8 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US
Station name None, Station number None
Caller ID Info Follows:
Standard BELLCORE
Voice card specific Info Follows:
Digit Duration Timing is set to 100 ms
```

The following table describes significant fields in the output from this command.

Table 29: show ephone-dn Field Descriptions

Field	Description
Administrative State	Administrative (configured) state of the voice port.
alert	The number of calls that were disconnected by the far-end device when the local IP phone was in the call alerting state (for example, because the far-end phone rang but was not answered and the far-end system decided to drop the call rather than let the phone ring for too long).
answered (incoming)	The number of incoming calls that were actually answered (the phone goes off hook when ringing).
answered (outgoing)	The number of outgoing call attempts that were answered by the far end.
busy	The number of outgoing call attempts that got a busy response.
Call Disconnect Time Out	Not applicable to the Cisco IP phone.
called, calling	Extension numbers of called and calling parties.
Caller ID Info Follows	Information about the caller ID.

Field	Description
Call Ref	A unique per-call identifier used by the SCCP protocol. The Call Ref values are assigned sequentially within the Cisco CME–SCCP interface, so this value also indicates the total number of SCCP calls since the router was last rebooted.
chan	Channel number of an ephone-dn.
CODEC	Codec type.
Companding Type	Not applicable to the Cisco IP phone.
connect	The number of calls that were disconnected by the far-end device when the local IP phone was in the call connected state.
Connection Mode	Not applicable to the Cisco IP phone.
Connection Number	Not applicable to the Cisco IP phone.
Description	Not applicable to the Cisco IP phone.
Digit Duration Timing	Not applicable to the Cisco IP phone.
DN STATE	Ephone-dn tag number and state of the phone line associated with an extension.
Echo Cancellation...	Not applicable to the Cisco IP phone.
Echo Cancel Coverage	Not applicable to the Cisco IP phone.
EFXS	Voice port type.
Far-end disconnect at...	See connect, alert, hold, and ring.
Final Jitter	The final voice packet receive jitter reported by the IP phone at the end of the call.
hold	The number of calls that were disconnected by the far-end device when the local IP phone was in the call hold state (for example, if the caller was left on hold for too long and got tired of waiting).
incoming	The number of incoming calls presented (the phone rings).
In Gain	Not applicable to the Cisco IP phone.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Last 64 far-end disconnect cause codes	See the Mappings of PSTN Cause Codes to SIP Event table for a list of public switch telephone network (PSTN) cause codes that can be sent as an ISDN cause information element (IE) and the corresponding Session Interface Protocol (SIP) event.

Field	Description
Latency	The final voice packet receive latency reported by the IP phone at the end of the call.
Lost	Number of lost packets.
Music On Hold Threshold	Not applicable to the Cisco IP phone.
No Interface Down Failure	State of the interface.
Noise Regeneration	Not applicable to the Cisco IP phone.
Non Linear...	Not applicable to the Cisco IP phone.
Operation State	Operational state of the voice port.
Out Attenuation	Not applicable to the Cisco IP phone.
outgoing	The number of outgoing call attempts.
Playout-delay Maximum	Not applicable to the Cisco IP phone.
Playout-delay...	Not applicable to the Cisco IP phone.
Port	Port number for the interface associated with the voice interface card.
Region Tone	Not applicable to the Cisco IP phone.
ring	The number of calls that were disconnected by the far-end device when the local IP phone was in the ringing state (for example, if the call was not answered and the caller hung up).
Ringing Time Out	Duration, in seconds, for which ringing is to continue if a call is not answered. Set with the **timeouts ringing** command.
Rx Pkts, bytes	Number of packets and bytes received during the current or last call.
Signal Level to phone, peak	For G.711 calls only, this parameter indicates the most recent voice signal level in the voice IP packets sent from the router to the IP phone. This parameter is valid only for VoIP or PSTN G.711 calls to the IP phones. This parameter is not valid for calls between local IP phones, or calls that use codecs other than G.711. The peak field indicates the peak signal level seen during the entire call.
Slot	Slot used in the voice interface card for this port.
Station name	Station name.
Station number	Station number.
Stream Port	RTP port allocated by the given DN/channel.
Sub-unit	Subunit used in the voice interface card for this port.

Field	Description
Tx Pkts, bytes	Number of packets and bytes transmitted during the current call or last call.
Type of VoicePort	Voice port type.
VAD	Voice activity detection.
Voice card specific info	Information specific to the voice card.
VPM STATE	State indication for the VPM software component.
VTSP STATE	State indication for the VTSP software component.
Wait Release Time Out	Time that a voice port stays in the call-failure state while the router sends a busy tone, reorder tone, or out-of-service tone to the port.

The following table lists the PSTN cause codes that can be sent as an ISDN cause information element (IE) and the corresponding SIP event for each. These are the far-end disconnect cause codes listed in the output for the **show ephone-dn statistics command.**

Table 30: Mappings of PSTN Cause Codes to SIP Events

PSTN Cause Code	Description	SIP Event
1	Unallocated number	410 Gone
3	No route to destination	404 Not found
16	Normal call clearing	BYE
17	User busy	486 Busy here
18	No user responding	480 Temporarily unavailable
19	No answer from the user	
21	Call rejected	603 Decline
22	Number changed	302 Moved temporarily
27	Destination out of order	404 Not found
28	Address incomplete	484 Address incomplete
29	Facility rejected	501 Not implemented
31	Normal unspecified	404 Not found

PSTN Cause Code	Description	SIP Event
34	No circuit available	503 Service unavailable
38	Network out of order	
41	Temporary failure	
42	Switching equipment congestion	
44	Requested channel not available	
47	Resource unavailable	
55	Incoming class barred within CUG	603 Decline
57	Bearer capability not authorized	501 Not implemented
58	Bearer capability not presently available	
63	Service or option unavailable	503 Service unavailable
65	Bearer cap not implemented	501 Not implemented
79	Service or option not implemented	
87	User not member of CUG	603 Decline
88	Incompatible destination	400 Bad Request
95	Invalid message	
102	Recover on timer expiry	408 Request timeout
111	Protocol error	400 Bad request
127	Interworking unspecified	500 Internal server error
Any code other than those listed above	500 Internal server error	

Related Commands

Command	Description
show ephone-dn callback	Displays information about pending callbacks in a Cisco Unified CME or a Cisco Unified SRST environment.
show ephone-dn loopback	Displays information about loopback ephone-dns that have been created in a Cisco Unified CME or a Cisco Unified SRST environment.
show ephone-dn statistics	Displays display call statistics for a Cisco IP destination or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

Command	Description
show ephone-dn summary	Displays brief information about Cisco IP phone destination numbers or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn callback

To display information about pending callbacks in a Cisco Unified CallManager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the **show ephone-dn callback** command in privileged EXEC mode.

show ephone-dn callback

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following sample output shows a callback placed by ephone-dn 1 against ephone-dn 3. Ephone-dn 3 has its channel 1 on hold and has just seized dial tone on its channel 2.

```
Router# show ephone-dn callback
DN 3 (95021) CallBack pending to DN 1 (95021) for ephone-1 age 7 seconds
State for DN 3 is CH1 HOLD      CH2 SIEZE
```

The following sample output shows a callback placed by ephone-dn 1 against ephone-dn 3. Ephone-dn 3 has a call in progress on channel 1.

```
Router# show ephone-dn callback
DN 3 (95021) CallBack pending to DN 1 (95021) for ephone-1 age 8 seconds
State for DN 3 is CH1 CONNECTED
```

Significant fields in the output from this command are described in the following table.

Table 31: show ephone-dn callback Field Descriptions

Field	Description
DN 3 (95021) CallBack pending to DN 1 (95021)	Callback originator is the extension with the dn-tag 1 (in this example), and the callback has been placed on the extension with the dn-tag 3 and the number 95021.
age	Number of seconds since the callback was placed.
State for DN 3 is CH1... CH2...	Call states for channel 1 and channel 2, if any, of the extension that the callback is for.

Related Commands

Command	Description
show ephone-dn	Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn conference

To display information about ad hoc and meet-me conferences in a Cisco Unified CallManager Express (Cisco Unified CME) environment, use the **show ephone-dn conference** command in privileged EXEC mode.

show ephone-dn conference [{**ad-hoc** [**video**] | **meetme** [**video**] | **number** *number*}]

Syntax Description

ad-hoc	(Optional) Displays adhoc conferences.
meetme	(Optional) Displays meet-me conferences.
video	(Optional) Displays video conferences.
number *number*	(Optional) Displays the conference telephone or extension number.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ2	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. The command output was enhanced to display the unlocked Meet-Me conference setting.
15.1(4)M	Cisco CME 8.6	This command was modified to display information on video conferences.

Examples

The following sample output displays information for the 1397 conference number. There are three directory numbers and six inactive parties. The number of unlocked DN tags are displayed at the end of each MeetMe conference.

```
Router# show ephone-dn conference number 1397
type   active inactive  numbers
==================================
Meetme    0       6      1397
DN tags: 10, 11, 12
Unlocked DN tags: 2/3
Meetme    0       4      2486
DN tags: 13, 14
All DN tags unlocked.
Meetme    0       4      1111
DN tags: 15, 16
Ad-hoc    0       4      7777
DN tags: 20, 21
Router# sh ephone-dn conference  ad-hoc video
type     active  inactive   numbers
================================================
Ad-hoc-video     3        3       2000
```

```
DN tags: 20, 21, 22
Router# sh ephone-dn conference  meetme video
type     active  inactive   numbers
================================================
Meetme-video     0        8       3000
1. DN tags: 25
```

The following table describes the significant fields shown in the display.

Table 32: show ephone-dn conference Field Descriptions

Field	Description
active	Number of active parties in the conference.
DN tags	Directory numbers (DNs) in the conference.
inactive	Number of inactive parties in the conference.
number	Conference telephone or extension number.
type	Type of conference: meet-me or ad hoc.

Related Commands

Command	Description
show ephone-dn	Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn loopback

To display information about loopback ephone-dns that have been created in a Cisco Unified CallManager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the **show ephone-dn loopback** command in privileged EXEC mode.

show ephone-dn loopback

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0 Cisco SRST 1.0	This command was introduced.
12.2(8)T	Cisco CME 2.0 Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Examples

The following example displays information for a loopback using ephone-dn 21 and ephone-dn 22:

```
Router# show ephone-dn loopback
LOOPBACK DN status (min 21, max 22):
DN 21 51...  Loopback to DN 22 CH1 IDLE
CallingDn -1 CalledDn -1 Called   Calling   G711Ulaw64k
Strip NONE, Forward 2, prefix 10 retry 10 Media 0.0.0.0 0
callID 0 srcCallID 0 ssrc 0 vector 0
DN 22 11...  Loopback to DN 21 CH1 IDLE
CallingDn -1 CalledDn -1 Called   Calling   G711Ulaw64k
Strip NONE, Forward 2, prefix 50 retry 10 Media 0.0.0.0 0
callID 0 srcCallID 0 ssrc 0 vector 0
```

Significant fields in the output from this command are described in the following table.

Table 33: show ephone-dn loopback Field Descriptions

Field	Description
Called, Calling	Called number and calling number when there is a call present.
CalledDn, CallingDn	Ephone-dn tag numbers of the called and calling ephone-dn. Set to -1 if the call is not to or from an ephone-dn, or if there is no active call.
callID	Internal call reference. This usage is the same as in other Cisco IOS voice gateway commands.
DN	Ephone-dn tag (sequence number).
Forward	Number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair.

Field	Description
G711...	G711Ulaw64k indicates G.711 codec, mu-law, 64000-bit stream. G711alaw64k indicates G.711 codec, A-law, 64000-bit stream.
Loopback to command...	Indicates the opposite ephone-dn in the loopback pair and the status of that ephone-dn.
Media	IP destination address, if any, for any voice packets that are passing through the loopback DN.
min, max	Lowest and highest dn-tag numbers of ephone-dns that are configured as loopback-dns.
prefix	Digit string to add to the beginning of forwarded called numbers.
retry	Number of seconds to wait before retrying the loopback target when is it busy.
srcCallID	Internal call reference for the destination.
ssrc	Real-time transport protocol (RTP) synchronization source (SSRC) of the most recent RTP packet.
Strip	Number of leading digits to strip before forwarding to the other extension in the loopback-dn pair.
vector	The following values describe the media path for voice packets that pass through the loopback-dn: • 0—No media path or not a loopback-dn path (inactive). • 1—Normal path. Loopback-dn has identified the final media destination as a local IP phone. The media IP address field shows a valid, non-zero value. • 2—Hairpin. Media packets are routed back through paired loopback-dns. The final destination is not known. For example, this can be a VoIP-to-VoIP call path by a loopback-dn. • 3—Hairpin. The final destination is an ephone-dn in a special mode such as paging. • 4—Loopback-dn chain has been detected, in which two loopback-dn pairs have been connected together. • 5—Loopback-dn chain has been detected in which more than two loopback-dn pairs are connected in series.

Related Commands

Command	Description
loopback-dn	Creates a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP voice calls and supplementary services.
show ephone-dn	Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn paging

To display configuration information on paging groups, use the **show ephone-dn paging** command in user EXEC or privileged EXEC mode.

show ephone-dn paging

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)

Privileged EXEC (#)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **show ephone-dn paging** command to display which paging dn is specified and which phone is being paged.

Examples

The following is a sample output from the **show ephone-dn paging** command before paging. The output shows two parts: the static "Paging Configuration" part and the dynamic "Paging Control Info" part. The output of the **show ephone-dn paging** command should be exactly the same before and after paging.

```
Router# show ephone-dn paging
Paging Configuration
ephone-dn 250 ( IDLE )
 number 7770
 paging ip 239.1.1.0 port 20480
   ephone-2[1]          paging-dn 250(OFF)
   ephone-7[6]          paging-dn 250(OFF)
 paging group 251,252
   voice reg pool 1     pagingGrp 251(OFF)
   voice reg pool 2     pagingGrp 252(OFF)
ephone-dn 251 ( IDLE )
 number 7771
 paging ip 239.1.1.1 port 20480
   voice reg pool 1     paging-dn 251(OFF)
ephone-dn 252 ( IDLE )
 number 7772
 paging ip 239.1.1.2 port 20480
   voice reg pool 2     paging-dn 252(OFF)
ephone-dn 253 ( IDLE )
 number 7773
 paging ip 239.1.1.3 port 20480
   ephone-8[7]          paging-dn 253(OFF)
        Paging Control Info
skinnyPC[0] ephone-paging-dn 250        ( IDLE ) count 0
skinnyPC[1] ephone-paging-dn 251        ( IDLE ) count 0
skinnyPC[2] ephone-paging-dn 252        ( IDLE ) count 0
skinnyPC[4] ephone-paging-dn 253        ( IDLE ) count 0
```

The following is a sample output from the **show ephone-dn paging** command during paging. In this output, the “Paging Configuration” part remains the same expect for the changes in state from IDLE to ACTIVE and OFF to ON. However, the “Paging Control Info” part displays the changes in the paging control information.

```
Router# show ephone-dn paging
        Paging Configuration
ephone-dn 250 (ACTIVE)
 number 7770
 paging ip 239.1.1.0 port 20480
   ephone-2[1]          paging-dn 250(ON )
   ephone-7[6]          paging-dn 250(OFF)
 paging group 251,252
   voice reg pool 1     pagingGrp 251(ON )
   voice reg pool 2     pagingGrp 252(ON )
ephone-dn 251 ( IDLE )
 number 7771
 paging ip 239.1.1.1 port 20480
   voice reg pool 1     paging-dn 251(ON )
ephone-dn 252 ( IDLE )
 number 7772
 paging ip 239.1.1.2 port 20480
   voice reg pool 2     paging-dn 252(ON )
ephone-dn 253 ( IDLE )
 number 7773
 paging ip 239.1.1.3 port 20480
   ephone-8[7]          paging-dn 253(OFF)
        Paging Control Info
skinnyPC[0] ephone-paging-dn 250        (ACTIVE) count 1
    phone           ip address      port
ephone#[phone]  2[1]            239.1.1.0   20480
sccp(ephone#[phone]): 2[1](mcast)
 group 251      (ephone#[phone]): None
 group 252      (ephone#[phone]): None
sip (pool[peer tag]): None
 group 251      (pool[peer tag]): 1[40001](mcast)
 group 252      (pool[peer tag]): 2[40003](mcast)
skinnyPC[1] ephone-paging-dn 251        ( IDLE ) count 0
skinnyPC[2] ephone-paging-dn 252        ( IDLE ) count 0
skinnyPC[4] ephone-paging-dn 253        ( IDLE ) count 0
```

The following is another sample output from the **show ephone-dn paging** command during paging:

```
Paging Configuration
ephone-dn 250 ( IDLE )
 number 7770
 paging ip 239.1.1.0 port 20480
 paging group 251
   ephone-2[1]          pagingGrp 251(ON )
   voice reg pool 3     pagingGrp 251(ON )
ephone-dn 251 (ACTIVE)
 number 7771
 paging ip 239.1.1.1 port 20480
   ephone-2[1]          paging-dn 251(ON )
   voice reg pool 3     paging-dn 251(ON )
        Paging Control Info
skinnyPC[0] ephone-paging-dn 250        ( IDLE ) count 0
skinnyPC[1] ephone-paging-dn 251        (ACTIVE) count 1
                phone           ip address      port
ephone#[phone]  2[1]            239.1.1.1       20480
sccp(ephone#[phone]): 2[1](m)
sip (pool[peer tag]): 3[40007](m)
```

The following table describes the significant fields shown in the display.

Table 34: show ephone-dn paging Field Descriptions

Field	Description
phone	Indicates the ephone-dn and the paging-dn tag.
ip address	Indicates the IP multicast address to multicast voice packets for audio paging.
port	Indicates the UDP port for multicast paging. Range is from 2000 to 65535. **Note** The correct paging port for the paging-dn of Cisco Unified SIP IP phones is an even number from 20480 to 32768 only.

Related Commands

Command	Description
paging-dn	Creates a paging extension (paging-dn) to receive audio pages on a Cisco Unified IP phone in a Cisco Unified CME system.
paging-dn (voice register)	Registers a Cisco Unified SIP IP phone to an ephone-dn paging directory number.
paging group	Creates a combined paging group from two or more previously established paging sets.

show ephone-dn park

To display information about call-park slots in the system, use the **show ephone-dn park** command in privileged EXEC mode.

show ephone-dn park

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(7)T	This command was introduced.

Examples

The following example shows information for a single call-park slot that uses an ephone-dn identifier of 50 and an extension number of 1560.

```
Router
#
 show ephone-dn park
DN 50 (1560) park-slot state IDLE
Notify to () timeout 15 limit 20
```

The following table describes the significant fields shown in the display.

Table 35: show ephone-dn park Field Descriptions

Field	Description
DN	Ephone-dn tag (identifier) number for the call-park slot.
(1560)	Extension number associated with the call-park slot.
park-slot state	Whether the call-park slot is in use or idle.
Notify to ( )	Extension that has been specified for notification. Empty parentheses indicate that no extension was specified in the configuration.
timeout	Number of seconds between reminder rings, in seconds.
limit	Number of reminder rings before a call parked at this slot is disconnected.

Related Commands

Command	Description
park-slot	Creates a floating extension (ephone-dn) at which calls can be temporarily held (parked).

show ephone-dn statistics

To display call statistics for a Cisco IP destination or for extensions (ephone-dns) in a Cisco Unified CallManager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the **show ephone-dn** command in privileged EXEC mode.

show ephone-dn [*dn-tag*] **statistics**

Syntax Description

dn-tag	(Optional) Unique sequence number that is used during configuration to identify a particular extension (ephone-dn).
statistics	Displays voice quality statistics on calls for a specified extension or for all extensions.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ1	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following sample output displays statistics for all extensions (ephone-dns) in a Cisco Unified CME system. There are two ephone-dns (DN1 and DN3) in this example.

```
Router# show ephone-dn statistics
Total Calls 103
Stats may appear to be inconsistent for conference or shared line cases
DN 1 chan 1 incoming 36 answered 21 outgoing 60 answered 30 busy 6
Far-end disconnect at:connect 29 alert 18 hold 7 ring 15
Last 64 far-end disconnect cause codes
17 17 17 17 17 17 16 16 16 16 16 16 16 16 16 16
16 16 16 16 65 16 65 65 65 65 16 65 65 65 16 16
16 16 16 16 16 16 16 16 16 16 16 16 16 65 47 65
47 47 16 16 16 16 16 16 16 16 16 16 16 16 16 16
local phone on-hook
DN 1 chan 1 (95011) voice quality statistics for last call
Call Ref 103 called 91500 calling 95011
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 30 Latency 0 Lost 0
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0
DN 1 chan 2 incoming 0 answered 0 outgoing 1 answered 0 busy 0
Far-end disconnect at:connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
local phone on-hook
DN 1 chan 2 (95011) voice quality statistics for last call
Call Ref 86 called  calling
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 0 Latency 0 Lost 0
```

```
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0
DN 3 chan 1 incoming 0 answered 0 outgoing 1 answered 1 busy 0
Far-end disconnect at:connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
DN 3 chan 1 (95021) voice quality statistics for current call
Call Ref 102 called 94011 calling 95021
Current Tx Pkts 241 bytes 3133 Rx Pkts 3304 bytes 515023 Lost 0
Jitter 30 Latency 0
Worst Jitter 30 Worst Latency 0
Signal Level to phone 201 (-39 dB) peak 5628 (-12 dB)
Packets counted by router 3305
```

The following sample output displays voice quality statistics for the ephone-dn with dn-tag 2:

```
Router# show ephone-dn 2 statistics
DN 2 chan 1 incoming 0 answered 0 outgoing 2 answered 0 busy 0
Far-end disconnect at: connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
28 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
local phone on-hook
DN 2 chan 1 (91450) voice quality statistics for last call
Call Ref 2 called  calling
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 0 Latency 0 Lost 0
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0
```

The **show ephone-dn** command describes significant fields in the output from this command.

Related Commands

Command	Description
show ephone-dn	Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn summary

To display brief information about Cisco IP phone destination numbers or for extensions (ephone-dns) in a Cisco Unified CallManager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the **show ephone-dn summary** command in privileged EXEC mode.

show ephone-dn summary

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0 Cisco SRST 1.0	This command was introduced.
12.2(8)T	Cisco CME 2.0 Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Examples

The following is example output from the **show ephone-dn summary**:

```
Router# show ephone-dn summary
PORT     DN STATE    CODEC    VAD VTSP STATE            VPM STATE
======== ==========  ======== === ===================== =========
50/0/1   DOWN        -         -  -                     EFXS_ONHOOK

50/0/2   DOWN        -         -  -                     EFXS_ONHOOK

50/0/3   DOWN        -         -  -                     EFXS_ONHOOK

50/0/4   INVALID     -         -  -                     EFXS_INIT

50/0/5   INVALID     -         -  -                     EFXS_INIT

50/0/6   INVALID     -         -  -                     EFXS_INIT
```

The following table describes significant fields in the output from this command.

Table 36: show ephone-dn summary Field Descriptions

Field	Description
CODEC	Type of codec.
DN STATE	Status of the ephone-dn.
EFXS	Voice port type.
PORT	Port number (virtual) for this interface. The number that follows the last slash in the port number is the ephone-dn tag. For example, if the port number is 50/0/1, the dn-tag is 1.
VAD	Voice activity detection status.

Field	Description
VPM STATE	State indication for the voice port module (VPM) software component.
VTSP STATE	State indication for the voice telephony service provider (VTSP) software component.

Related Commands

Command	Description
show ephone-dn	Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn whisper

To display information about whisper intercom ephone-dns that have been created in Cisco Unified CME, use the **show ephone-dn whisper** command in privileged EXEC mode.

show ephone-dn whisper

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Examples

The following is sample output from the **show ephone-dn whisper** command showing an active whisper intercom call between extension 6001 and 6002:

```
Router# show ephone-dn whisper

DN        DN NUMBER     LABEL         SPEED DIAL    DN STATE       PHONE
==        =========     =====         ==========    ========       =====
101       8881          wi_8881       -               IDLE         35 w36
102       8882          -             -               IDLE         36
103       8883          wi_8883       -               IDLE         m35 38
104       8884          wi_8884       -               IDLE         38
105       8885          wi_8885_sd_8888882            IDLE
106       8886          wi_8886_sd_8888883            IDLE         36
107       8887          -             8888            IDLE         35
108       8888          Mary_sd_Peter 8887            IDLE         36
109       8889          -             -               IDLE
110       8890          wi_8890       -               IDLE
111       4441          4441_wi_sd_4444442            IDLE
112       4442          wi_4442       -               IDLE
113       4443          -             -               IDLE
114       4444          4444_sd-8882  8882            IDLE
141       5551          -             -               IDLE
142       5552          -             -               IDLE
143       5553          -             -               IDLE
144       5554          -             -               IDLE
145       5555          -             -               IDLE
161       6001          -             6002            WHISPER      1
162       6002          -             6001            WHISPER      2
163       6003          -             6001            IDLE
164       6004          -             6002            IDLE
166       6006          -             6003            IDLE
167       6007          -             6003            IDLE
168       6008          -             6002            IDLE
169       6009          -             6006            IDLE
```

The following table describes the significant fields in the output from this command in alphabetical order.

Table 37: show ephone-dn whisper Field Descriptions

Field	Description
DN	Directory number tag.
DN Number	Extension or telephone number assigned to directory number.
Label	Text string that identifies the whisper intercom line.
Speed Dial	Whisper intercom number to speed dial.
DN State	State of the directory number, either Idle or Busy.
Phone	Ephone that the directory number is assigned to.

Related Commands

Command	Description
debug ephone whisper-intercom	Displays debugging messages for the Whisper Intercom feature.
show ephone-dn	Displays status and configuration information for phone extensions (ephone-dns) in Cisco Unified CME.
whisper-intercom	Enables the Whisper Intercom feature on a directory number.

show ephone-hunt

To display ephone-hunt configuration information and current status and statistics information, use the **show ephone-hunt** command in privileged EXEC mode.

show ephone-hunt [{*tag* | **summary**}]

Syntax Description

tag	(Optional) Hunt-group number that was used to identify a hunt group in the **ephone-hunt** command. Range is 1 to 100.
summary	(Optional) Displays hunt group configuration information.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The **show ephone-hunt** and **show ephone-hunt summary** commands display information for peer, sequential, and last-idle ephone hunt groups. Using the *tag* argument outputs data for a specific ephone hunt group.

The output is dependent on call activity. If there is no activity, no data is displayed.

Examples

The following examples are contained in this section:

Verbose Output

The following is a sample output from the **show ephone-hunt** command when no argument or keyword has been entered. The sample contains information for a peer hunt group, a sequential hunt group, and a longest-idle hunt group. See the table for descriptions of significant fields in the output.

```
Router# show ephone-hunt
Group 1
    type: peer
    pilot number: 450, peer-tag 20123
    list of numbers:
        451, aux-number A450A0900, # peers 5, logout 0, down 1
            peer-tag  dn-tag  rna  login/logout  up/down
             [20122     42     0       login      up  ]
             [20121     41     0       login      up  ]
             [20120     40     0       login      up  ]
             [20119     30     0       login      up  ]
             [20118     29     0       login      down]
        452, aux-number A450A0901, # peers 4, logout 0, down 0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20127     45     0       login      up  ]
             [20126     44     0       login      up  ]
             [20125     43     0       login      up  ]
             [20124     31     0       login      up  ]
```

```
        453, aux-number A450A0902, # peers 4, logout 0, down 0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20131     48     0       login      up  ]
             [20130     47     0       login      up  ]
             [20129     46     0       login      up  ]
             [20128     32     0       login      up  ]
        477, aux-number A450A0903, # peers 1, logout 0, down 0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20132     499    0       login      up  ]
    preference: 0
    members initial state: logout
    preference (sec): 7
    timeout: 3, 3, 3, 3
    max timeout : 10
    hops: 4
    next-to-pick: 1
    E.164 register: yes
    auto logout: no
    stat collect: no
Group 2
    type: sequential
    pilot number: 601, peer-tag 20098
    list of numbers:
        123, aux-number A601A0200, # peers 1, logout 0, down 0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20097     56     0       login      up  ]
        622, aux-number A601A0201, # peers 3, logout 0, down 0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20101     112    0       login      up  ]
             [20100     111    0       login      up  ]
             [20099     110    0       login      up  ]
        623, aux-number A601A0202, # peers 3, logout 0, down 0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20104     122    0       login      up  ]
             [20103     121    0       login      up  ]
             [20102     120    0       login      up  ]
        *, aux-number A601A0203, # peers 1, logout 0, down 1
            peer-tag  dn-tag  rna  login/logout  up/down
             [20105     0      0       -          down]
        *, aux-number A601A0204, # peers 1, logout 0, down 1
            peer-tag  dn-tag  rna  login/logout  up/down
             [20106     0      0       -          down]
    final number: 5255348
    preference: 0
    members initial state: logout
    preference (sec): 9
    timeout: 5, 5, 5, 5, 5
    max timeout : 40
    fwd-final: orig-phone
    E.164 register: yes
    auto logout: no
    stat collect: no
Group 3
    type: longest-idle
    pilot number: 100, peer-tag 20142
    list of numbers:
        101, aux-number A100A9700, # peers 3, logout 0, down 3
            on-hook time stamp 7616, off-hook agents=0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20141     132    0       login      down]
             [20140     131    0       login      down]
             [20139     130    0       login      down]
        *, aux-number A100A9701, # peers 1, logout 0, down 1
            on-hook time stamp 7616, off-hook agents=0
```

```
            peer-tag  dn-tag  rna  login/logout  up/down
             [20143     0      0       -          down]
        102, aux-number A100A9702, # peers 2, logout 0, down 2
            on-hook time stamp 7616, off-hook agents=0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20145     142    0       login      down]
             [20144     141    0       login      down]
    all agents down!
    preference: 0
    members initial state: logout
    preference (sec): 7
    timeout: 100, 100, 100
    hops: 0
    E.164 register: yes
    auto logout: no
    stat collect: no
```

Summary Output

The following example shows a summary output. See the table for descriptions of significant fields in the output.

```
Router# show ephone-hunt summary
Group 1
    type: peer
    pilot number: 5000
    list of numbers:
       5001
       5002
       5003
       5004
       5005
    final number: 5006
    preference: 0
    members initial state: logout
    timeout: 180
    hops: 2
    E.164 register: yes
Group 2
    type: sequential
    pilot number: 6000
    list of numbers:
       5005
       5004
       5003
       5002
       5001
    final number: 5007
    preference: 5
    members initial state: logout
    timeout: 3
    E.164 register: no
```

Agent Status Control Conditions

A portion of the **show ephone-hunt** command output displays the ready and not-ready agent status of extensions in hunt groups. An extension that is ready is available to receive hunt-group calls. An extension that is in not-ready status blocks hunt-group calls. An agent toggles an extension from ready to not ready and back to ready using the HLog soft key or a FAC.

The following examples display some output that reports different agent status not-ready conditions within a hunt group. In the hunt group used for these examples, there are four users: agent1 and agent4 share extension 8001, agent2 is on extension 8002, and agent3 is on extension 8003.

In the **show ephone-hunt** output, "logout 0" means that all instances of the extension are in ready status. Any number greater than zero next to "logout' indicates that at least one ephone using the extension has activated not-ready status.

If agent1 is in not-ready status, the **show ephone-hunt** command will display the following output. The logout value for extension 8001 is 1 because one phone is in not-ready status.

```
Router# show ephone-hunt
.
.
.
list of numbers:
 8001, aux-number A8000A100, # peers 2, logout 1 ...
 8002, aux-number A8000A101, # peers 1, logout 0...
 8003, aux-number A8000A102, # peers 1, logout 0...
.
```

If agent1 and agent2 place their phones in not-ready status, the **show ephone-hunt** command will display the following output:

```
Router# show ephone-hunt
.
.
.
list of numbers:
    8001, aux-number A8000A100, # peers 2, logout 1...
    8002, aux-number A8000A101, # peers 1, logout 1...
    8003, aux-number A8000A102, # peers 1, logout 0...
```

If all agents place their phones in not-ready status, the **show ephone-hunt** command displays the following output. Note that the logout value of 2 for extension 8001 indicates that both ephone-dns with that extension number (agent1 and agent4) are in not-ready status.

```
Router# show ephone-hunt
.
.
.
list of numbers:
   8001, aux-number A8000A100, # peers 2, logout 2...
   8002, aux-number A8000A101, # peers 1, logout 1...
   8003, aux-number A8000A102, # peers 1, logout 1...
all agents logout!
```

Automatic Agent Status Not-Ready Parameters

The **show ephone-hunt** command displays the parameters that have been set using the **auto logout** command, which is used for the Automatic Agent Status Not-Ready feature. The table shows the possible values of the auto logout field. describes other fields in the output.

```
Router# show ephone-hunt 1
```

```
Group 1
   type:sequential
   pilot number:8888, peer-tag 20029
   list of numbers:
       8001, aux-number A8888A000, # peers 1, logout 0, down 0
           peer-tag:dn-tag [ 20028:1]
       8003, aux-number A8888A001, # peers 1, logout 0, down 0
           peer-tag:dn-tag [ 20030:3]
   preference:0
   members initial state: logout
   preference (sec):9
   timeout:5
   E.164 register:yes
   auto logout:no
   stat collect:yes
```

Table 38: show ephone-hunt Auto Logout Examples

show ephone-hunt Output	**Description**	**auto logout Command**
`auto logout: no`	The Automatic Agent Status Not-Ready feature is disabled. This is also the default if this command is not used.	`no auto logout`
`auto logout: 1 type: both`	The Automatic Agent Status Not-Ready feature is enabled and no options have been used with the **auto logout** command. The number of unanswered calls is 1 and the command applies to both static and dynamic hunt group members by default.	`auto logout`
`auto logout: 2 type: both`	Two unanswered calls will be sent to a hunt group agent before the agent's status is automatically changed to not ready. The command applies to both static and dynamic hunt group members by default.	`auto logout 2`
`auto logout: 3 type: static`	Three unanswered calls will be sent to a hunt group agent before the agent's status is automatically changed to not ready. The command applies to static hunt group members only.	`auto logout 3 static`

The table describes significant fields shown in **show ephone-hunt** command displays.

Table 39: show ephone-hunt Field Descriptions

Field	**Description**
auto logout	Indicates whether the Automatic Agent Status Not-Ready feature has been enabled. See the table.
aux-number	Auxiliary number used to generate dial peers for a hunt group. This number is generated by the **list** command.
description	Description string entered for the ephone hunt group. This value is set using the **description (ephone-hunt)** command.
dn-tag	Directory number (DN) sequence number.
E.164 register	Displays whether a pilot number registers with an H.323 gatekeeper. This value is set by the **no-reg** command.

Field	Description
final number	Last number in the ephone-hunt group, after which a call is no longer redirected. This value is set by the **final** command.
fwd-final	Final destination of an unanswered call that has been transferred into a hunt group: orig-phone means calls are returned to the transferring phone, and final means calls are sent to the final number specified in the configuration. This value is set by the **fwd-final** command.
hops	Number of hops before a call proceeds to the final number. This value is set by the **hops** command.
list of numbers	Extension numbers that are group members of the specified ephone hunt group. This value is set by the **list** command.
login/logout	Ready status of the agent: login means ready and accepting calls, and logout means not-ready and blocking hunt-group calls.
logout	Number of agents in the not-ready state (not accepting hunt-group calls).
max timeout	Maximum combined timeout for the no-answer periods for all ephone-dns in the ephone-hunt list. This value is set by the **max-timeout** command.
members initial state: logout/login	Sets all static members initial state to logout.
next-to-pick	(Peer hunt groups only) List number of the agent whose phone will ring when the next call comes in to the hunt group. (For example, if the order of agents in the **list** command is 451, 452, 453, 454, the list number 2 represents extension 452.)
off-hook agents	Number of agents who are currently off-hook.
on-hook time stamp	(Longest-idle hunt groups only) The last on-hook time of the agent, which is used to determine which agent to ring next time.
peers	Displays the number of ephone-dn dial peers.
peer-tag	Dial-peer sequence number.
pilot number	Number that callers dial to reach the ephone hunt group.
preference	Preference order set by the **preference (ephone-hunt)** command for the primary pilot number.
preference (sec)	Preference order set by the **preference (ephone-hunt)** command for the secondary pilot number.
rna	Number of unanswered hunt group calls (ring-no-answer) by this agent, used for the Automatic Agent Status Not-Ready feature.
stat collect	Indicates whether statistic are being Cisco Unified CME B-ACD data is being collected. See the **statistics collect** command.

Field	Description
timeout	Number of seconds after which a call that is not answered at one number is redirected to the next number in the hunt-group list. Multiple values in this field refer to the timeouts for the hops between ephone-dns in a hunt group as they appear in the **list** command. This value is set by the **timeout** command.
type	Type of ephone hunt group: longest-idle, peer, or sequential.
up/down	Dial peer is up or down.

Related Commands

Command	Description
auto logout	Enables automatic change of agent status to not-ready after a specified number of hunt-group calls are not answered.
ephone-hunt	Enters ephone-hunt configuration mode to create a hunt group for use in a Cisco Unified CME system.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.
members logout	Sets all static members initial state to logout.
show ephone-hunt statistics	Displays hunt group call statistics.

show ephone-hunt statistics

To display ephone-hunt statistics information, use the **show ephone-hunt statistics** command in privileged EXEC mode.

show **ephone-hunt** *tag* **statistics** {**last** *hours* **hours** | **start** *day* *time* [**to** *day* *time*]}

Syntax Description

tag	Hunt-tag number that was used to identify a hunt group in an **ephone-hunt** command. Range is 1 to 100.
last	Displays information for the previous number of specified hours, counting backward from the current hour. Range is 1 to 167.
hours **hours**	Number of hours for which to display call statistics.
start	Defines the start of a period for which to display call statistics. Default duration is one hour.
day	Day of week. Use **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, or **sat**.
time	Hour of day. Range is 0 to 23.
to	(Optional) Defines the stop time for display of call statistics.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	Call hold statistics were added.
12.4(9)T	Cisco Unified CME 4.0	Call hold statistics were integrated into Cisco IOS Release 12.4(9)T.
15.2(2)T	Cisco Unified CME 9.0	This command was modified to add the following fields: Calls handoff to IOS, Average time to handoff, Longest time to handoff, and Number of error calls.

Usage Guidelines

The **show ephone-hunt** statistics last and **show ephone-hunt** statistics commands provide expanded information regarding extension (list of numbers) and pilot numbers.

The output is dependent on call activity. If there is no activity, no data is displayed.

If your Cisco Unified CME system is configured with the basic automatic call distribution (B-ACD) and auto-attendant service, you can enable the collection of call statistics per ephone hunt group with the **statistics collect** command. Additional data is displayed for all agents combined and for individual agents. The additional data includes statistics such as: the number of calls received, the amount of time the calls waited to be answered, and the amount of time the calls spent on hold or in a queue.

The **statistics collect** command can be used to obtain other call statistics, such as direct calls to hunt group pilot numbers. For more information, see the " Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant Service " chapter in the *Cisco Unified CME B-ACD and TCL Call-Handling Applications* guide.

Once you have enabled statistics collection, you can use the **show ephone-hunt statistics** command to display call statistics, or you can use the **hunt-group report every hours** and **hunt-group report url** commands to transfer the statistics to files using TFTP.

Note On the day that daylight saving time adjusts the time back by one hour at 2 a.m. each year, the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

Examples

The following is a sample output that displays call statistics for the past hour for hunt group 2, which is associated with a Cisco Unified CME B-ACD service:

```
Router# show ephone-hunt 2 stat last 1 h
Thu 02:00 - 03:00
    Max Agents: 3
    Min Agents: 3
    Total Calls: 9
    Answered Calls: 7
    Abandoned Calls: 2
    Average Time to Answer (secs): 6
    Longest Time to Answer (secs): 13
    Average Time in Call (secs): 75
    Longest Time in Call (secs): 161
    Average Time before Abandon (secs): 8
    Calls on Hold: 2
    Average Time in Hold (secs): 16
    Longest Time in Hold (secs): 21
    Per agent statistics:
      Agent: 8004
        From Direct Call:
          Total Calls Answered : 3:
          Average Time in Call (secs) : 70
          Longest Time in Call (secs) : 150
          Total Calls on Hold : 1:
          Average Hold Time (secs) : 21
          Longest Hold Time (secs) : 21
        From Queue:
          Total Calls Answered : 3
          Average Time in Call (secs) : 55
          Longest Time in Call (secs) : 78
          Total Calls on Hold : 2:
          Average Hold Time (secs) : 19
          Longest Hold Time (secs) : 26
      Agent: 8006
        From Direct Call:
          Total Calls Answered : 3:
          Average Time in Call (secs) : 51
          Longest Time in Call (secs) : 118
          Total Calls on Hold : 1:
          Average Hold Time (secs) : 11
          Longest Hold Time (secs) : 11
        From Queue:
          Total Calls Answered : 1
          Average Time in Call (secs) : 4
```

```
          Longest Time in Call (secs) : 4
      Agent: 8044
        From Direct Call:
          Total Calls Answered : 1:
          Average Time in Call (secs) : 161
          Longest Time in Call (secs) : 161
        From Queue:
          Total Calls Answered : 1
          Average Time in Call (secs) : 658
          Longest Time in Call (secs) : 658
Queue related statistics:
      Total calls presented to the queue:  5
      Calls handoff to IOS: 2
      Number of calls in the queue: 1
      Average time to handoff (secs):  2
      Longest time to handoff (secs):  3
      Number of abandoned calls:  0
      Average time before abandon (secs):  0
      Calls forwarded to voice mail:  0
      Calls answered by voice mail:  0
Number of error calls: 0
```

The following is a sample output from the **show ephone-hunt statistics** command. The output focuses on queue-related statistics.

```
Queue related statistics:
      Total calls presented to the queue: 8
      Calls handoff to IOS: 3
Number of calls in the queue: 1
      Average time to handoff (secs): 10
      Longest time to handoff (secs): 15
Number of abandoned calls: 4
      Average time before abandon (secs): 7
      Calls forwarded to voice mail: 0
      Calls answered by voice mail: 0
      Number of error calls: 0
```

The table describes the significant fields shown in the output of the **show ephone-hunt statistics** command, in alphabetical order.

Table 40: show ephone-hunt statistics Field Descriptions

Field	Description
Abandoned calls	Total number of calls abandoned by hunt group agents. This does not include calls going to the final number.
Answered call	Total number of calls answered by hunt group agents.
Average time before abandon (secs)	Average length of time that unanswered calls waited before hanging up.
Average hold time (secs)	Average length of time that calls waited on hold for this agent.
Average time in call (secs)	Average length of time that unanswered calls waited before going to an agent.
Average time in hold (secs)	Average length of time that calls were kept on hold for all agents.

Field	Description
Average time to answer (secs)	Average length of time that all calls to Cisco Unified CME B-ACD waited before being answered.
Average time to handoff (secs)	Average length of time before a call was handed off to IOS.
Calls answered by voice mail	Total number of calls to Cisco Unified CME B-ACD that were answered by voice mail.
Calls exited the queue	Total number of calls to Cisco Unified CME B-ACD that exited queues.
Calls forwarded to voice mail	Total number of calls to Cisco Unified CME B-ACD that were forwarded to voice mail.
Calls handoff to IOS	Total number of calls handed off to IOS.
Calls on hold	Total number of calls that were placed on hold.
Longest hold time (secs)	Longest length of time that a call to this agent spent between being placed on hold and being picked up.
Longest time in call (secs)	Longest length of time in which calls to Cisco Unified CME B-ACD went to an agent and waited in a call queue.
Longest time in hold (secs)	Longest length of time that a call spent between being placed on hold and being picked up by agents.
Longest time to answer (secs)	Longest length of time before calls to Cisco Unified CME B-ACD were answered.
Longest time to handoff (secs)	Longest length of time before a call was handed off to IOS.
Max agent	Maximum number of hunt group agents.
Min agent	Minimum number of hunt group agents.
Number of abandoned calls:	Total number of calls to Cisco Unified CME B-ACD that hung up before being answered.
Number of error calls	Total number of misdialed calls.
Total calls answered	Total number of calls to Cisco Unified CME B-ACD that were answered by an agent.
Total calls on hold	Total number of calls that were on hold for this agent.
Total calls presented to the queue	Total number of calls made to Cisco Unified CME B-ACD.
Total calls	Total number of direct calls made to the hunt group.

> **Note** From Cisco Unified CME Release 10.5 onwards, abandoned calls will not include the calls going to the final number. However, the total calls includes calls going to the final number. Use the formula " **Final Calls= Total Calls - Answered Calls - Abandoned Calls**" , to calculate the calls going to the final number.

Related Commands

Command	Description
ephone-hunt	Enters ephone-hunt configuration mode to create a hunt group for use in a Cisco Unified CME system.
hunt-group report every hours	Sets the hourly interval at which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco Unified CME B-ACD call statistics are to be sent using TFTP.
statistics collect	Enables the collection of call statistics for an ephone hunt group.

show fb-its-log

To display information about the Cisco CallManager Express (Cisco CME) eXtensible Markup Language (XML) application program interface (API) configuration, statistics on XML API queries, and the XML API event logs, use the **show fb-its-log** command in privileged EXEC mode.

show fb-its-log [**summary**]

Syntax Description

summary	(Optional) Displays only the XML API configuration and the statistics for queries and logs, and not the logs themselves.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(15)ZJ	3.0	This command was introduced.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following is sample output from the **show fb-its-log** summary command:

```
Router# show fb-its-log summary
IP Keyswitch Logs:(21:11:30 UTC Wed Jul 1 2003)
  ---- Current Period ---
    extension events:4
    device events: 3
    overwrites:0
    missed:0
    deleted:0
 ---- History ------
    overwrites:0
    missed:0
    deleted:8
---- Threads ----
    max xml threads:2
    current thread:0
    read in process:FALSE
```

The following is sample output from the **show fb-its-log** command:

```
Router# show fb-its-log
IP Keyswitch Logs:(21:11:30 UTC Wed Jul 1 2003)
  ---- Current Period ---
    extension events:4
    device events: 3
    overwrites:0
    missed:0
    deleted:0
 ---- History ------
    overwrites:0
    missed:0
    deleted:8
---- Threads ----
    max xml threads:2
```

```
    cuttent thread:0
    read in process:FALSE
1  Time:21:11:06 UTC Wed Jul 1 2003
    Event:DN 1[2001] goes down
2  Time:21:11:06 UTC Wed Jul 1 2003
    Event:DN 2[2003] goes down
3  Time:21:11:06 UTC Wed Jul 1 2003
    Event:IP Phone 1[SEP003094C3F96A] unregistered
4  Time:21:11:06 UTC Wed Jul 1 2003
    Event:IP Phone 1[SEP003094C3F96A] unregistered
5  Time:21:11:54 UTC Wed Jul 2003
    Event:IP Phone 1[SEP003094C3F96A] registered
6  Time:21:11:57 UTC Wed Jul 2003
    Event:DN 1[2001] goes up
7  Time:21:11:57 UTC Wed Jul 2003
    Event:DN 2[2003] goes up
```

The following table describes the significant fields in this output.

Table 41: show fb-its-log Field Descriptions

Field	Description
Current Period	The time between the last retain-timer-triggered cleanup to the next cleanup.
extension events	Events related to extensions that have been captured in the internal buffer.
device events	Events related to devices that have been captured in the internal buffer.
overwrites	Events that are written over previously recorded events in the buffer. Overwrites occur when the internal buffer size is too small; new events overwrite old ones. The internal buffer size is set using the **max-size** keyword in the **log table** command.
missed	Events that happen too quickly for the system to record.
deleted	Events removed from the internal buffer.
History	Information since the last system restart.
Threads	Current number of threads configured in the system.
max xml threads	Maximum number of concurrent XML threads allowed.
current thread	XML API query thread.
read in process	TRUE indicates that the xml-test.html file is being read now. FALSE indicates that the file is not being read.
UTC	Coordinated Universal Time, which is used by the system clock on the Cisco CME router.

Related Commands

Command	Description
log table	Sets the maximum size of the table used to capture phone events used for the Cisco CME XML API.

show ip address trusted list

To display a list of trusted ip addresses, use the **show ip address trusted list** command in privileged EXEC mode.

show ip address trusted list

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use this command to display a list of trusted IP addresses.

Examples

The following is a sample output from this command displaying all statistical information:

```
Router #show ip address trusted list
IP Address Trusted Authentication
 Administration State: UP
 Operation State:      UP
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 Session Targets:
Peer Tag        Oper State      Session Target
--------        ----------      --------------
11              DOWN            ipv4:1.3.45.1
1               UP              ipv4:1.3.45.1
IP Address Trusted List:
 ipv4 172.19.245.1
 ipv4 172.19.247.1
 ipv4 172.19.243.1
 ipv4 171.19.245.1
 ipv4 171.19.10.1
```

Related Commands

Command	Description
ip address trusted list	Allows to add a list of trusted IP addresses.

show presence global

To display configuration information about the presence service, use the **show presence global** command in user EXEC or privileged EXEC mode.

show presence global

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command displays the configuration settings for presence.

Examples

The following example displays output from the **show subscription global**:

```
Router# show subscription global
Presence Global Configuration Information:
=============================================
Presence feature enable              : TRUE
Presence allow external watchers     : FALSE
Presence max subscription allowed    : 100
Presence number of subscriptions     : 0
Presence allow external subscribe    : FALSE
Presence call list enable            : TRUE
Presence server IP address           : 0.0.0.0
Presence sccp blfsd retry interval   : 60
Presence sccp blfsd retry limit      : 10
Presence router mode                 : CME mode
```

The table describes the significant fields shown in the display.

Table 42: show subscription global Field Descriptions

Field	Description
Presence feature enable	Indicates whether presence is enabled on the router with the **presence** command.
Presence allow external watchers	Indicates whether internal presentities can be watched by external watchers, as set by the **watcher all**
Presence max subscription allowed	Maximum number of presence subscriptions allowed by the **max-subscription** command.
Presence number of subscriptions	Current number of active presence subscriptions.

Field	Description
Presence allow external subscribe	Indicates whether internal watchers are allowed to subscribe to status notifications from external presentities, as set by the **allow subscribe** command.
Presence call list enable	Indicates whether the Busy Lamp Field (BLF) call-list feature is enabled with the **presence call-list** command.
Presence server IP address	Displays the IP address of an external presence server defined with the **server** command.
Presence sccp blfsd retry interval	Retry timeout, in seconds, for BLF speed-dial numbers on SCCP phones set by the **sccp blf-speed-dial retry interval** command.
Presence sccp blfsd retry limit	Maximum number of retries allowed for BLF speed-dial numbers on SCCP phones set by the **sccp blf-speed-dial retry interval** command.
Presence router mode	Indicates whether the configuration mode is set to Cisco Unified CME or Cisco Unified SRST by the **mode** command.

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
allow subscribe	Allows internal watchers to monitor external presence entities (directory numbers).
debug presence	Displays debugging information about the presence service.
presence enable	Allows the router to accept incoming presence requests.
server	Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.
show presence subscription	Displays information about active presence subscriptions.
watcher all	Allows external watchers to monitor internal presence entities (directory numbers).

show presence subscription

To display information about active presence subscriptions, use the **show presence subscription** command in user EXEC or privileged EXEC mode.

show presence subscription [{**details** | **presentity** *telephone-number* | **subid** *subscription-id* | **summary**}]

Syntax Description

details	(Optional) Displays detailed information about presentities, watchers, and presence subscriptions.
presentity *telephone-number*	(Optional) Displays information on the presentity specified by the destination telephone number.
subid *subscription-id*	(Optional) Displays information for the specific subscription ID.
summary	(Optional) Displays summary information about active subscription requests.

Command Default

Information for all active presence subscriptions is displayed.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(24)T	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command displays details about the currently active presence subscriptions

Examples

The following is sample output from the **show presence subscription details** command:

```
Presence Active Subscription Records Details:
=============================================

Subscription ID         : 1
  Watcher               : 6002@10.4.171.60
  Presentity            : 6005@10.4.171.34
  Expires               : 3600 seconds
  Subscription Duration : 1751 seconds
  line status           : idle
  watcher type          : local
  presentity type       : local
  Watcher phone type    : SIP Phone
  subscription type     : Incoming Indication
  retry limit           : 0
  sibling subID         : 0
  sdb                   : 0
  dp                    : 6555346C
  watcher dial peer tag : 40001
```

```
  number of presentity  : 1

Subscription ID         : 2
  Watcher               : 6002@10.4.171.60

Presence Active Subscription Records:
=============================================

Subscription ID         : 30
  Watcher               : 4085550103@10.4.171.34
  Presentity            : 5001@10.4.171.20
  Expires               : 3600 seconds
  line status           : idle
  watcher type          : local
  presentity type       : remote
  Watcher phone type    : SCCP [BLF Call List]
  subscription type     : Outgoing Request
  retry limit           : 0
  sibling subID         : 23
  sdb                   : 0
  dp                    : 0
  watcher dial peer tag : 0
```

The following is sample output from the **show presence subscription summary** command:

```
Router# show presence subscription summary

Presence Active Subscription Records Summary: 15 subscription
Watcher                   Presentity               SubID  Expires SibID  Status
======================== ======================== ====== ======= ====== ======
6002@10.4.171.60          6005@10.4.171.34          1      3600    0      idle
6005@10.4.171.81          6002@10.4.171.34          6      3600    0      idle
6005@10.4.171.81          6003@10.4.171.34          8      3600    0      idle
6005@10.4.171.81          6002@10.4.171.34          9      3600    0      idle
6005@10.4.171.81          6003@10.4.171.34          10     3600    0      idle
6005@10.4.171.81          6001@10.4.171.34          12     3600    0      idle
6001@10.4.171.61          6003@10.4.171.34          15     3600    0      idle
6001@10.4.171.61          6002@10.4.171.34          17     3600    0      idle
6003@10.4.171.59          6003@10.4.171.34          19     3600    0      idle
6003@10.4.171.59          6002@10.4.171.34          21     3600    0      idle
6003@10.4.171.59          5001@10.4.171.34          23     3600    24     idle
6002@10.4.171.60          6003@10.4.171.34          121    3600    0      idle
6002@10.4.171.60          5002@10.4.171.34          128    3600    129    idle
6005@10.4.171.81          1001@10.4.171.34          130    3600    131    busy
6005@10.4.171.81          7005@10.4.171.34          132    3600    133    idle
```

The following is sample output from the **show presence subscription summary** command showing that device-based BLF monitoring is enabled on two phones.

```
Watcher                     Presentity                SubID  Expires SibID  Status
========================== ======================== ====== ======= ====== ======
D 2036@10.6.2.6             2038@10.6.2.254           33     3600    0      idle

  2036@10.6.2.6             2038@10.6.2.254           35     3600    0      idle

D 2036@10.6.2.6             8883@10.6.2.254           37     3600    0      unknown
```

The following is sample output from the **show presence subscription subid** command:

```
Router# show presence subscription subid 133

Presence Active Subscription Records:
=============================================

Subscription ID         : 133
  Watcher               : 6005@10.4.171.34
  Presentity            : 7005@10.4.171.20
  Expires               : 3600 seconds
  line status           : idle
  watcher type          : local
  presentity type       : remote
  Watcher phone type    : SIP Phone
  subscription type     : Outgoing Request
  retry limit           : 0
  sibling subID         : 132
  sdb                   : 0
  dp                    : 0
  watcher dial peer tag : 0
```

The following table describes the significant fields shown in the display.

Table 43: show presence subscription Field Descriptions

Field	Description
Watcher	IP address of the watcher.
Presentity	IP address of the presentity.
Expires	Number of seconds until the subscription expires. Default is 3600.
line status	Status of the line: • Idle—Line is not being used. • In-use—User is on the line, whether or not this line can accept a new call. • Unknown—Phone is unregistered or this line is not allowed to be watched.
watcher type	Whether the watcher is local or remote.
presentity type	Whether the presentity is local or remote.
Watcher phone type	Type of phone, either SCCP or SIP.
subscription type	The type of presence subscription, either incoming or outgoing.
retry limit	Maximum number of times the router attempts to subscribe for the line status of an external SCCP phone when either the presentity does not exist or the router receives a terminated NOTIFY from the external presence server. Set with the **sccp blf-speed-dial retry-interval** command.
sibling subID	Sibling subscription ID if presentity is remote. If value is 0, presentity is local.
sdb	Voice port of the presentity.
dp	Dial peer of the presentity.

Field	Description
watcher dial peer tag	Dial peer tag of the watcher device.

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
blf-speed-dial	Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.
debug ephone blf	Displays debugging information for BLF presence features.
debug presence	Displays debugging information about the presence service.
presence	Enables presence service and enters presence configuration mode.
presence enable	Allows the router to accept incoming presence requests.
show presence global	Displays configuration information about the presence service.

show sdspfarm

To display the status of the configured digital signal processor (DSP) farms and transcoding streams, use the **show sdspfarm** command in privileged EXEC mode.

show sdspfarm {**units** | **sessions** {**active** | **callID** *number* | **statistics** | **summary**}}

Syntax Description

units	Displays the configured and registered DSP farms.
sessions	Displays the transcoding streams.
active	Displays all active sessions.
callID	Displays activities for a specific caller ID.
number	Displays caller ID number displayed by the **show voip rtp connection** command.
statistics	Displays session statistics.
summary	Displays summary information.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Products	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.

Examples

The following is sample output from the **show sdspfarm units** :

```
Router# show sdspfarm units
mtp-1 Device:MTP123456782012 TCP socket:[-1]  UNREGISTERED
actual_stream:0 max_stream 0 IP:0.0.0.0  0  Unknown 0 keepalive 0
mtp-2 Device:MTP000a8aeaca80 TCP socket:[5]  REGISTERED
actual_stream:40 max_stream 40 IP:10.5.49.160  11001  MTP YOKO keepalive 12074
Supported codec:G711Ulaw
                 G711Alaw
                 G729
                 G729a
                 G729b
                 G729ab
 max-mtps:2, max-streams:240, alloc-streams:40, act-streams:0
The following is sample output from the show sdspfarm sessions active
:
Router# show sdspfarm sessions active
Stream-ID:3 mtp:2 1.5.49.160  20174  Local:2000 START
 usage:MoH  (DN=3  , CH=1) FE=TRUE
 codec:G729  duration:20 vad:0 peer Stream-ID:4
Stream-ID:4 mtp:2 1.5.49.160  17072  Local:2000 START
 usage:MoH  (DN=3  , CH=1) FE=FALSE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:3
```

The following is sample output from the **show sdspfarm sessions callID** :

```
Router# show sdspfarm sessions callid 51M
Stream-ID:6, srcCall-ID:51, codec:G729AnnexA , dur:20ms, vad:0, dstCall-ID:52, confID:5,
mtp:2^
Peer Stream-ID:5, srcCall-ID:52, codec:G711Ulaw64k , dur:20ms, vad:0, dstCall-ID:51, confID:5,
 mtp:2^
Router-2015# show sdspfarm sessions callid 52
Stream-ID:5, srcCall-ID:52, codec:G711Ulaw64k , dur:20ms, vad:0, dstCall-ID:51, confID:5,
mtp:2
Peer Stream-ID:6, srcCall-ID:51, codec:G729AnnexA , dur:20ms, vad:0, dstCall-ID:52, confID:5,
 mtp:2
```

The following is sample output from the **show sdspfarm sessions statistics** :

```
Router# show sdspfarm sessions statistics
Stream-ID:1 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:1014 in-pak:0 discard:0
Stream-ID:2 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:3 mtp:2 10.5.49.160  20174  Local:2000START  MoH  (DN=3  , CH=1) FE=TRUE
 codec:G729  duration:20 vad:0 peer Stream-ID:4
 recv-pak:0 xmit-pak:0 out-pak:4780 in-pak:0 discard:0
Stream-ID:4 mtp:2 10.5.49.160  17072  Local:2000START  MoH  (DN=3  , CH=1) FE=FALSE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:3
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:5 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:6 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:7 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:8 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:9 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:10 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:11 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:12 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:13 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:14 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:15 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:16 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:17 mtp:2 0.0.0.0  0  Local:0IDLE
```

```
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:18 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:19 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:20 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:21 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:22 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:23 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:24 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:25 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:26 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:27 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:28 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:29 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:30 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:31 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:32 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:33 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:34 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:35 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:36 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:37 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:38 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
```

```
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:39 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:40 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
```

The following is sample output from the **show sdspfarm sessions summary** :

```
Router# show sdspfarm sessions summary
max-mtps:2, max-streams:240, alloc-streams:40, act-streams:2
  ID   MTP  State      CallID confID Usage                         Codec/Duration
==== ===== ====== =========== ====== ============================= ==============
1    2     IDLE   -1          0                                   G711Ulaw64k /20ms
2    2     IDLE   -1          0                                   G711Ulaw64k /20ms
3    2     START  -1          3      MoH  (DN=3  , CH=1) FE=TRUE  G729 /20ms
4    2     START  -1          3      MoH  (DN=3  , CH=1) FE=FALSE G711Ulaw64k /20ms
5    2     IDLE   -1          0                                   G711Ulaw64k /20ms
6    2     IDLE   -1          0                                   G711Ulaw64k /20ms
7    2     IDLE   -1          0                                   G711Ulaw64k /20ms
8    2     IDLE   -1          0                                   G711Ulaw64k /20ms
9    2     IDLE   -1          0                                   G711Ulaw64k /20ms
10   2     IDLE   -1          0                                   G711Ulaw64k /20ms
11   2     IDLE   -1          0                                   G711Ulaw64k /20ms
12   2     IDLE   -1          0                                   G711Ulaw64k /20ms
13   2     IDLE   -1          0                                   G711Ulaw64k /20ms
14   2     IDLE   -1          0                                   G711Ulaw64k /20ms
15   2     IDLE   -1          0                                   G711Ulaw64k /20ms
16   2     IDLE   -1          0                                   G711Ulaw64k /20ms
17   2     IDLE   -1          0                                   G711Ulaw64k /20ms
18   2     IDLE   -1          0                                   G711Ulaw64k /20ms
19   2     IDLE   -1          0                                   G711Ulaw64k /20ms
20   2     IDLE   -1          0                                   G711Ulaw64k /20ms
21   2     IDLE   -1          0                                   G711Ulaw64k /20ms
22   2     IDLE   -1          0                                   G711Ulaw64k /20ms
23   2     IDLE   -1          0                                   G711Ulaw64k /20ms
24   2     IDLE   -1          0                                   G711Ulaw64k /20ms
25   2     IDLE   -1          0                                   G711Ulaw64k /20ms
26   2     IDLE   -1          0                                   G711Ulaw64k /20ms
27   2     IDLE   -1          0                                   G711Ulaw64k /20ms
28   2     IDLE   -1          0                                   G711Ulaw64k /20ms
29   2     IDLE   -1          0                                   G711Ulaw64k /20ms
30   2     IDLE   -1          0                                   G711Ulaw64k /20ms
31   2     IDLE   -1          0                                   G711Ulaw64k /20ms
32   2     IDLE   -1          0                                   G711Ulaw64k /20ms
33   2     IDLE   -1          0                                   G711Ulaw64k /20ms
34   2     IDLE   -1          0                                   G711Ulaw64k /20ms
35   2     IDLE   -1          0                                   G711Ulaw64k /20ms
36   2     IDLE   -1          0                                   G711Ulaw64k /20ms
37   2     IDLE   -1          0                                   G711Ulaw64k /20ms
38   2     IDLE   -1          0                                   G711Ulaw64k /20ms
39   2     IDLE   -1          0                                   G711Ulaw64k /20ms
40   2     IDLE   -1          0                                   G711Ulaw64k /20ms
```

The following table describes the fields shown in the show sdspfarm display.

Table 44: show sdspfarm Field Descriptions

Field	Description
act-streams	Active streams that are currently involved in calls.

Field	Description
alloc-streams	Number of transcoding streams that are actually allocated to all DSP farms that are registered to Cisco CME.
callID	Caller ID that the active stream is in.
Codec	Codec in use.
confID	ConfID that is used to communicate with DSP farms.
discard	Number of packets that are discarded.
dstCall-ID	Caller ID of the destination IP call leg.
Duration or dur	Packet rates, in milliseconds.
ID	Transcoding stream sequence number in Cisco CME.
in-pak	Number of incoming packets from the source call leg.
Local	Local port for voice packets.
max-mtps	Maximum number of Message Transfer Parts (MTPs) that are currently allowed to register in Cisco CME.
max-streams	Maximum number of transcoding streams that are currently allowed in Cisco CME.
mtp or MTP	MTP sequence number where the transcoding stream is located.
out-pak	Number of outgoing packets sending to source call leg.
peer Stream-ID	Stream sequence number of the other stream paired in the same transcoding session. (Two transcoding streams make up a transcoding session).
recv-pak	Number of voice packets received from DSP farm.
srcCall-ID	Source caller ID of the source IP call leg.
State	Current state of the transcoding stream, could be IDLE, SEIZE, START, STOP, or END.
Stream-ID	Transcoding stream sequence number in Cisco CME.
TCP-socket	Socket number for DSP farm (similar to TCP socket for **show ephone** output).
usage	Current usage of the stream; for example, Ip-Ip (IP to IP transcoding), MOH (for MOH transcoding) and Conf (conference).
vad	Voice-activity detection (VAD) flag for the transcoding stream. It should always be 0 (False).
xmit-pak	Number of packets that are sent to DSP farm.

Related Commands

Command	Description
sdspfarm tag	Permits a DSP farm to be to registered to Cisco CME and be associated with an SCCP client interface's MAC address.
sdspfarm transcode sessions	Specifies the maximum number of transcoding sessions allowed per Cisco CME router.
sdspfarm units	Specifies the maximum number of DSP farms that are allowed to be registered to Cisco CME.

show shared-line

To display information about the Session Initiation Protocol (SIP) shared lines, use the **show shared-line** command in user EXEC or privileged EXEC mode.

show shared-line {**call** | **details** | **subscription** | **summary**}

Syntax Description

call	Displays information about all active calls on shared lines.
details	Displays detailed information about each shared line.
subscription	Displays information for specific subscriptions to shared lines.
summary	Displays summary information about active subscriptions to shared lines.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Release	Modification
12.4(24)T	This command was introduced.

Examples

The following is sample output from the **show shared-line call** command:

```
Router# show shared-line call
Shared-Line active call info:
Shared-Line: '20141',  active calls: 3
Local User       Local Address                Remote User      Remote Address   CallID
==========       =====================        ===========      ===================== ====
20141            20141@10.6.0.2                20143            20143@10.10.0.1  3168

20141            20141@10.6.0.1                Barge            20143@10.10.0.1  3209

20141            20141@10.6.0.2                20141            20141@10.10.0.1  3210
```

The following is sample output from the **show shared-line details** command:

```
Router# show shared-line details
Shared-Line info details:

Shared-Line: '20141',  subscribed users: 2,  max calls limit: 10
Index        Users                  sub_id        peer_tag        Status
=====        =====                  ======        ========        ======
  1          20141@10.6.0.1            5             40001           ACTIVE
  2          20141@10.6.0.2            6             40002           ACTIVE
Free call queue size: 7,   Active call queue size: 3

Message queue size: 20,   Event queue size: 64
```

The following is sample output from the **show shared-line subscription** command:

```
Router# show shared-line subscription
```

```
Shared-Line Subscription Info:

Subscriptions to: '20141',   total subscriptions: 2
SubID        Subscriber                   Expires         Sub-Status
=====        ==========                   =======         ==========
  5          20141@10.6.0.1                  3600          NOTIFY_ACKED
  6          20141@10.6.0.2                  3600          NOTIFY_ACKED
```

The following is sample output from the **show shared-line summary** command:

```
Router# show shared-line summary
Shared-Line info summary:
Shared-Line: '20141',  subscribed users: 2,  max calls limit: 10
```

The following table describes the significant fields shown in the displays.

Table 45: show shared-line Field Descriptions

Field	Description
Expires	Number of seconds until the subscription expires.
Local Address	IP address of the local phone involved in the shared line call.
Local User	Extension number of the shared line.
Remote Address	IP address of the remote phone involved in the shared line call.
Remote User	Extension of the remote phone involved in the shared line call.
SubID	Subscription ID.
Subscriber	Extension number of the shared line and the IP address of the phone subscriber.
Sub-Status	Status of the subscription.
Users	IP addresses of the phones using the shared line.

Related Commands

Command	Description
debug shared-line	Displays debugging information about SIP shared lines.

show telephony-service admin

To display information about the Cisco CallManager Express (Cisco CME) system administrator, use the **show telephony-service admin** command in user EXEC or privileged EXEC mode.

show telephony-service admin

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Examples

The following is sample output from this command:

```
Router# show telephony-service admin
admin_username Admin
admin_password word
edit DN through Web: enabled.
edit TIME through Web: enabled.
```

The following table describes the significant fields in this output.

Table 46: show telephony-service admin Field Descriptions

Field	Description
admin_username	Username of system administrator.
admin_password	Password of system administrator.
edit DN through Web	Whether editing of extensions through the GUI has been enabled using the **dn-webedit** command.
edit TIME through Web	Whether changing the router time through the GUI has been enabled using the **time-webedit** command.

Related Commands

Command	Description
dn-webedit	Enables adding of extensions (ephone-dns) through the web interface.

Command	Description
time-webedit	Enables setting of time through the web interface.

show telephony-service all

To display detailed configuration for phones, voice ports, and dial peers in a Cisco Unified Communications Manager Express (Cisco Unified CME) system, use the **show telephony-service all** command in user EXEC or privileged EXEC mode.

show telephony-service all

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	Cisco Unified CME 1.0	This command was introduced.
12.2(8)T	Cisco Unified CME 2.0	This command was integrated int Cisco IOS Release 12.2(8)T.
15.2(2)T	Cisco Unified CME 9.0	This command was modified to display the total number of data collected from both ephone and voice hunt groups.

Usage Guidelines

Use the **show telephony-service all** command to display the total number of ephone and voice hunt groups that have statistics collection turned on.

Examples

The following is a sample output from the **show telephony-service all** command:

```
Router# show telephony-service all
CONFIG
======
ip source-address 10.0.0.1 port 2000
max-ephones 24
max-dn 24
dialplan-pattern 1 408734....
voicemail 11111
transfer-pattern 510734....
keepalive 30
ephone-dn 1
number 5001
huntstop
ephone-dn 2
number 5002
huntstop
call-forward noan 5001 timeout 8
ephone-dn 3
number 5003
huntstop
ephone 1
mac-address 0030.94C3.37CB
type 0
button  1:1
speed-dial 1 5002
speed-dial 2 5003
cos 0
```

```
!
ephone 2
mac-address 0030.94C3.F96A
type 0
button  1:2 2:3 3:4
speed-dial 1 5004
speed-dial 2 5001
cos 0
!
voice-port 50/0/1
 station-id number 5001
!
voice-port 50/0/2
 station-id number 5002
 timeout ringing 8
!
dial-peer voice 20025 pots
 destination-pattern 5001
 huntstop
 port 50/0/1
dial-peer voice 20026 pots
 destination-pattern 5002
 huntstop
 call-forward noan 5001
 port 50/0/2
dial-peer voice 20027 pots
 destination-pattern 5003
 huntstop
 port 50/0/3
```

The following is a sample output from the **show telephony-service all** command. The output shows that call statistics are collected for 14 hunt groups, including 6 ephone and 8 voice hunt groups.

```
Router# show telephony-service all
CONFIG (Version=8.7)
=====================
Version 8.7
Max phoneload sccp version 17
Max dspfarm sccp version 18
Cisco Unified Communications Manager Express
For on-line documentation please see:
http://www.cisco.com/en/US/products/sw/voicesw/ps4625/tsd_products_support_series_home.html
protocol mode default
ip source-address 1.4.190.80 port 2000
ip qos dscp:
 ef (the MS 6 bits, 46, in ToS, 0xB8) for media
 cs3 (the MS 6 bits, 24, in ToS, 0x60) for signal
 af41 (the MS 6 bits, 34, in ToS, 0x88) for video
 default (the MS 6 bits, 0, in ToS, 0x0) for serviceservice directed-pickup
load 6921 SCCP69xx.9-0-3-0
load 6961 SCCP69xx.8-5-3-0
max-ephones 14
max-dn 56
max-conferences 4 gain -6
dspfarm units 0
dspfarm transcode sessions 0
conference software
privacy
no privacy-on-hold
hunt-group report url prefix tftp://223.255.254.254/ngm/huntgp/uc500/test
hunt-group report url suffix 0 to 20
hunt-group report every 1 hours
# of hunt-group collect data: 14
```

```
hunt-group report delay 0 hours
Number of ephone hunt-group configured: 6
hunt-group logout DND
max-redirect 20
cnf-file location: system:
cnf-file option: PER-PHONE-TYPE
```

The following is another sample output from the **show telephony-service all** command. The output shows that call statistics are collected for seven hunt groups, including three ephone and four voice hunt groups.

```
Router# show telephony-service all
.
.
.
.
.
hunt-group report url prefix tftp://223.255.254.254/ngm/huntgp/uc500/test
hunt-group report url suffix 0 to 20
hunt-group report every 1 hours
# of hunt-group collect data: 7
hunt-group report delay 0 hours
Number of ephone hunt-group configured: 3
.
.
.
.
.
```

The following table describes significant fields in this output, in alphabetical order.

Table 47: show telephony-service all Field Descriptions

Field	Description
button	Button on the Cisco IP phone.
call-forward noan	Call forward no answer is set.
cnf-file location	Storage location for phone configuration files. System (default), flash or slot 0 memory, and external TFTP server.
cnf-file option	Specifies the use of different phone configuration files by type of phone or by individual phone.
cos	Not applicable; unused.
destination-pattern	Destination pattern (telephone number) configured for this dial peer.
dial-peer voice	Voice dial peer.
dialplan-pattern	Dial-plan pattern is set to expand the abbreviated extension numbers to fully qualified E.164 numbers.
ephone	Cisco IP phone.
ephone-dn	Cisco IP phone directory number.
huntstop	Huntstop is set.

Field	Description
ip source-address	IP address used by Cisco IP phones to register with the router for service.
keepalive	IP phone keepalive period, in seconds.
mac-address	MAC address.
max-dn	Maximum directory numbers.
max-ephones	Maximum numbers of Cisco IP phones.
number	Cisco IP phone number.
port	TCP port number used by Cisco IP phones to communicate with the router.
pots	POTS dial peer set.
speed-dial	Speed-dial is set.
station-id number	Number used for caller ID purposes when calls are made using the line.
timeout	Timeout is set.
timeout ringing	Maximum amount of time that the phone is allowed to ring before the call is disconnected.
transfer-pattern	Transfer pattern is set to allow transfer of calls to a specified number.
type	Not applicable; unused.
voicemail	A voice-mail (speed-dial) number is set.
voice-port	(Virtual) voice port designator.
# of hunt-group collect data	Total number of data collected from both ephone and voice hunt groups.

Related Commands

Command	Description
show telephony dial-peer	Displays dial peers for extensions in a Cisco Unified CME system.
show telephony voice-port	Displays virtual voice-port configuration for extensions in a Cisco Unified CME system.

show telephony-service bulk-speed-dial

To display information about bulk speed-dial lists, use the **show telephony-service bulk-speed-dial** command in privileged EXEC mode.

show telephony-service bulk-speed-dial {**global** *list-id index-id* [**all**] | **local** *phone-tag list-id index-id* [**all**] | **summary**}

Syntax Description

global	Global lists that can be accessed by all users.
local	Personal lists that can be accessed by users configured to use the lists.
list-id	Digit that identifies the list. Range is from 0 to 9.
index-id	Identification number for an entry.
phone-tag	Ephone identifier (phone-tag).
summary	List of registered bulk speed-dial text files.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Examples

The following example displays the list of bulk speed-dial text files that have been configured in the system:

```
Router# show telephony-service bulk-speed-dial summary
List-id    Entries      Size    Reference  url
      0         40      3840    Global     tftp://192.168.254.254/phonedirs/uut.csv
      1         20      1920    Global     phoneBook.csv
      8         15      1440    Global     tftp://192.168.254.254/phonedirs/big.txt
      9         20      1920    Global     tftp://192.168.254.254/phonedirs/phoneBook.csv
      6      24879   2388384    ephone-2   tftp://192.168.254.254/phonedirs/big.txt1
      7         20      1920    ephone-2   phoneBook.csv
      6      24879   2388384    ephone-3   big.txt1
      7         20      1920    ephone-3   phoneBook.csv
4 Global List(s) 4 Local List(s)
```

The following example displays the single entry 1234 from list 9:

```
Router# show telephony-service bulk-speed-dial global 9 1234
Number: 1800 200 1345 name: Jay Smith Private: yes Extension: No
```

The following example displays all index entries starting with 1 for personal list number 7 for ephone 2:

```
Router# show telephony-service bulk-speed-dial local 2 7 1 all
```

```
Index  Number                  Name               Hide    Append
1000   918005550164            ABC Co Front Desk    no      no
1003   919005550167            ABC Co File room     no      no
1100   918005550118                                 no      no
1200   918005550184            ABC Co President     no      no
1301   918005550152                                 no      no
1342   91800,5550185           ABC Co Sales         no      no
1682   91800555,,0115          ABC Co Service       no      no
```

The following table describes the significant fields shown in the display.

Table 48: show telephony-service bulk-speed-dial Field Descriptions

Field	Description
List-id	Digit that identifies the list. Range is from 0 to 9.
Entries	Number of entries in the speed-dial file.
Size	Size of the file, in KB.
Reference	Assignment of the list: global if assigned to all ephones, or a specific ephone number.
url	Location of the text file, in URL format.
Index	Identification number for an entry.
Number	Number to be dialed and displayed on the phone.
Name	Name to be displayed on the phone.
Hide	Yes indicates that this number should not be displayed when it is dialed.
Append	Yes indicates that additional digits can be dialed by the user after this number has been speed-dialed before the call is completed.

Related Commands

Command	Description
bulk-speed-dial list (ephone)	Enables a personal bulk speed-dial list for an ephone.
bulk-speed-dial list (telephony-service)	Enables a global bulk speed-dial list for all users of a Cisco Unified CME system.
bulk-speed-dial prefix	Sets the prefix code that phone users dial to access speed-dial numbers from a bulk speed-dial list.

show telephony-service conference hardware

To display information about hardware conferences in a Cisco Unified Communications Manager Express (Cisco CME) system, use the **show telephony-service conference hardware** command in privileged EXEC mode.

show telephony-service conference hardware [{**ad-hoc** [{**detail** | **video**}] | **detail** [**video**] | **meetme** [{**detail** | **video**}] | **number** *telephone-number*}]

Syntax Description

ad-hoc	(Optional) Ad-hoc hardware conferences.
detail	(Optional) Detailed information for all conferences.
video	(Optional) Video conferences.
meetme	(Optional) Meet-me hardware conferences.
number	(Optional) Conference number.
telephone-number	(Optional) Telephone or extension number.

Command Modes

Privileged EXEC (#)

Command History

Release	Cisco Product	Modification
12.4(11)XJ2	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
15.1(4)M	Cisco Unified CME 8.6	This command was modified to include the video option.
15.2(2)T	Cisco Unified CME9.0	This command was modified to add hardware conference information on Cisco Unified SIP IP phones to the output display.

Usage Guidelines

Use the **show telephony-service conference hardware** command to display ad-hoc and meet-me hardware conferences information, including which parties are still in the conference.

Examples

The following is a sample output that displays information for a four-party ad-hoc hardware conference. Extension 8044 created the conference by calling extension 8012, then adding extension 8004 to the conference. The conference administrator, extension 8006, called into the conference after it was established.

```
Router# show telephony-service conference hardware detail
```

Conference Type Active Max Peak Host HostPhone Last

cur(initial)

===

8893 Ad-hoc 4 8 4 8044 29 (29) 8006

Conference parties:

8006 (admin)

8004

8012

```
    8044
```

The following is a sample output that displays information for a meet-me video conference:

```
Router# show telephony-service conference hardware detail video
Conference   Type              Active  Max  Peak    Host         HostPhone   Last
                                                                   cur(initial)

================================================================================
=============
9999         Meetme-Video          10    16    10   n/a            0      (  0)   9012

Conference parties (number:phone)
    9012  :12 :Audio
    7001  :Video
    9003  :3 :Audio
    7047  :Audio
    7015  :Video
    3667  :Audio
    9024  :24 :Audio
    9023  :23 :Video
    3665  :Video
    9022  :22 :Video
```

The following is another sample output from the **show telephony-service conference hardware detail** command. The output shows an ad-hoc video hardware conference among three participants, two of which are Cisco Unified SIP IP phones.

```
Router# show telephony-service conference hardware detail
Conference   Type Active  Max  Peak  Host                 HostPhone Last

           cur(initial)
========================================================================
B000             Ad-hoc Video    3         4       3        3915 SIP3915     15(15)
       5801 RM5801
Conference parties (number:phone)
   5801 5801 :Video
   3916 SIPPHONE3916 :16 :Video
   3915 SIPPHONE3915 :15  (admin):Video
```

The following is a sample output from the **show telephony-service conference hardware ad-hoc** command:

```
Router# show telephony-service conference hardware ad-hoc
Conference   Type Active  Max  Peak  Host                 HostPhone   Last

            cur(initial)
========================================================================
B000             Ad-hoc Video    3         4       3        3915 SIP3915     15(15)
       5801 RM5801
```

The following is a sample output from the **show telephony-service conference hardware meetme** command:

```
Router# show telephony-service conference hardware meetme
Conference   Type               Active  Max  Peak    Host                HostPhone   Last

                cur(initial)
========================================================================
7788            Meetme Video     4        4       4         3916 SIP3916    16(16)
      5802 RM5802
```

The following is a sample output from the **show telephony-service conference hardware number** command:

```
Router# show telephony-service conference hardware number B000
Conference   Type                Active  Max  Peak  Host                 HostPhone   Last

           cur(initial)
========================================================================
B000             Ad-hoc Video    3         4       3        3915 SIP3915     15(15)
       5801 RM5801
```

The following is another sample output from the **show telephony-service conference hardware number** command:

```
Router# show telephony-service conference hardware number 7788
Conference   Type               Active   Max  Peak  Host                HostPhone   Last

              cur(initial)
========================================================================
7788             Meetme Video   4         4        4       3917 SIP3917    17(17)
      4801  SCCP4801
```

The following table describes the significant fields shown in the display, listed in alphabetical order.

Table 49: show telephony-service conference hardware Field Descriptions

Field	Description
Active	Number of active parties in the conference.
admin	Ad hoc and meet-me hardware conference administrator. The administrator can: • Dial in to any conference directly through the conference number. • Use the ConfList soft key to list conference parties. • Remove any party from any conference.
Conference	Conference directory number (DN).
Conference parties	DNs in the conference.
Last	Last participant to join the conference.
Host	Conference creator.

Field	Description
HostPhone cur(initial)	cur—Current host phone. The phone that currently hosts the conference creator. (initial)—Initial host phone. The phone that hosted the conference creator when the conference was created. Because you can transfer the conference creator, the current host phone may be different from the initial host phone.
Max	Maximum number of participants allowed in the conference.
Peak	Maximum number of participants in the conference at any time.
Type	Type of conference: meet-me or ad hoc.

show telephony-service directory-entry

To display the entries made using the **directory entry** , use the **show telephony-service directory-entry** command in user EXEC or privileged EXEC mode.

show telephony-service directory-entry

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(15)ZJ	3.0	This command was introduced.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command lists directory entries that are made using the **directory entry** but does not list entries that are made using the **name** and **number** commands in ephone-dn configuration mode.

Examples

The following is sample output from this command:

```
Router# show telephony-service directory-entry
directory entry 1 4085550123 name Smith, John
```

The following table describes significant fields in this output, in alphabetical order.

Table 50: show telephony-service directory-entry Field Descriptions

Field	Description
directory *directory-tag* (shown as 1 in the example)	Sequence number or unique identifier for a directory entry.
name (shown as Smith, John)	Name that appears in the directory associated with the number.
number (shown as 4085550123 in the example)	Telephone number or extension for the directory entry.

Related Commands

Command	Description
directory entry	Adds an entry to a local phone directory that can be displayed on IP phones.
show telephony-service all	Displays detailed configuration of a Cisco CME system.
show telephony-service ephone-dn	Displays information for extensions (ephone-dns) in a Cisco CME system.

show telephony-service ephone

To display configuration for the Cisco IP phones, use the **show telephony-service ephone** command in user EXEC or privileged EXEC mode.

show telephony-service ephone

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0	This command was introduced.
12.2(8)T	Cisco CME 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The conference add-mode, conference drop-mode, and conference admin fields were added.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. The output was enhanced to include the setting of the **feature-button** command and information about logical partitioning class of restriction (LPCOR).
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Examples

The following is sample output from this command:

```
Router# show telephony-service ephone

Number of Configured ephones 2 (Registered 2)
ephone 1
Device Security Mode: Non-Secure
mac-address 1234.4321.7000
type 7960
button  1:1
keepalive 30 auxiliary 30
multicast-moh
max-calls-per-button 8
busy-trigger-per-button 0
Always send media packets to this router: No
Preferred codec: g711ulaw
conference drop-mode never
conference add-mode all
conference admin: No
privacy: Yes
feature-button 1 Dnd
user-locale US
```

```
network-locale US
lpcor type: remote
lpcor (incoming): ephone_group2  (outgoing): ephone_group2
!
ephone 2
Device Security Mode: Non-Secure
mac-address 1234.4321.6000
type 7960
button  1:2
keepalive 30 auxiliary 30
multicast-moh
max-calls-per-button 8
busy-trigger-per-button 0
Always send media packets to this router: No
Preferred codec: g711ulaw
conference drop-mode never
conference add-mode all
conference admin: No
privacy: Yes
feature-button 1 Dnd
user-locale US
network-locale US
lpcor type: local
lpcor (incoming): ephone_group1  (outgoing): ephone_group1
!
```

The table describes significant fields in this output, in alphabetical order.

Table 51: show telephony-service ephone Field Descriptions

Field	Description
button	Button number on IP phone, separator to denote ring characteristics and ephone-dn tag. A colon (:) separator denotes a normal ring.
conference add-mode	Who can add parties to a conference: • creator—Only the creator can add parties. • all—Any party can add other parties if the creator remains in the conference.
conference drop-mode	When conferences are dropped: • creator—Conference terminates when the creator hangs up. • local—Conference terminates when the last local party in the conference hangs up or drops out of the conference. • never—Conference is not dropped, even if the creator hangs up, as long as three parties remain in the conference.
conference admin	Ad hoc and meet-me hardware conference administrator. The administrator can: • Dial in to any conference directly through the conference number • Use the ConfList soft key to list conference parties • Remove any party from any conference
ephone	Cisco IP phone.
feature-button	Displays the type of feature button on the ephone. Feature type can be configured with privacy or DND.

Field	Description
lpcor (incoming)	Setting of the **lpcor incoming** command.
lpcor (outgoing)	Setting of the **lpcor outgoing** command.
lpcor type	Setting of the **lpcor type** command.
mac-address	MAC address of the Cisco IP phone.
speed-dial	Speed-tag (unique identifier) and the number that is programmed for that speed-tag.
type	Model type of phone.

Related Commands

Command	Description
show telephony-service all	Displays detailed configuration for a Cisco Unified CME system.
show telephony-service dial-peer	Displays dial-peer information for extensions in Cisco Unified CME.
show telephony-service ephone-dn	Displays information for extensions in Cisco Unified CME.
show telephony-service voice-port	Displays configurations for virtual voice ports in Cisco Unified CME.

show telephony-service ephone-dn

To display information about extensions (ephone-dns) in a Cisco CallManager Express (Cisco CME) system, use the **show telephony-service ephone-dn** command in user EXEC or privileged EXEC mode.

show telephony-service ephone-dn

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	1.0	This command was introduced
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Examples

The following is sample output from this command:

```
Router# show telephony-service ephone-dn
ephone-dn 1
 number 5001
 huntstop
ephone-dn 2
 number 5002
 huntstop
 call-forward noan 5001 timeout 8
ephone-dn 3
 number 5003
 huntstop
ephone-dn 4
 number 5004
 huntstop
```

The following table describes significant fields in this output, in alphabetical order.

Table 52: show telephony-service ephone-dn Field Descriptions

Field	Description
call-forward noan	Call forwarding is set to no answer. Other available options are call-forward busy and call-forward all.
ephone-dn	Cisco IP phone directory number.
huntstop	Huntstop is set.
number	Cisco IP phone number.
timeout	Timeout setting for call forwarding when an extension does not answer.

Related Commands

Command	Description
show telephony-service all	Displays the detailed configuration of all the Cisco IP phones.
show telephony-service dial-peer	Displays dial peer information for extensions (ephone-dns) in a Cisco CME system.
show telephony-service voice-port	Displays configurations for virtual voice ports in a Cisco CME system.

show telephony-service ephone-dn-template

To display information about ephone-dn-template configurations, use the **show telephony-service ephone-dn-template** command in user EXEC or privileged EXEC mode.

show telephony-service ephone-dn-template

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command displays contents of ephone-dn templates. Use the **show running-config** to display the association of templates to particular ephone-dns.

Examples

The following is sample output from this command:

```
Router# show telephony-service ephone-dn-template
ephone-template 1
 softkeys idle  Newcall Redial Cfwdall Dnd Pickup Gpickup Login
 codec g711ulaw
 User Locale: US
 Network Locale: US
ephone-template 2
 softkeys idle  Redial Newcall Dnd Cfwdall Pickup Gpickup Login
 codec g711ulaw
 User Locale: US
 Network Locale: US
```

Related Commands

Command	Description
ephone-dn-template	Creates an ephone-dn template and enters ephone-dn-template configuration mode.

show telephony-service ephone-template

To display the contents of ephone-templates, use the **show telephony-service ephone-template** command in user EXEC or privileged EXEC mode.

show telephony-service ephone-template

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(11)XJ2	Cisco Unified CME 4.1	The conference add-mode, conference drop-mode, and conference admin fields were added.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(15)XY	Cisco Unified CME 4.2(1)	Emergency response location (ERL) information assigned to an ephone displays in the output.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. Logical partitioning class of restriction (LPCOR) information was added to the output.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use this command to display the contents of each ephone template that is defined. Use the **show running-config** command to display the association of templates to specific ephones.

Examples

The following is sample output from this command:

```
Router# show telephony-service ephone-template

ephone-template 1
softkey idle  Cfwdall Dnd Gpickup Join Pickup RmLstC
softkey connected  Acct ConfList Confrn Endcall Hold Join Park
conference drop-mode never
conference add-mode all
conference admin: No
max-calls-per-button 8
busy-trigger-per-button 0
privacy default
MLPP max precedence level -1
MLPP indication Enabled
```

```
MLPP preemption Enabled
Always send media packets to this router: No
Preferred codec: g711ulaw
keepalive 30 auxiliary 30
User Locale: US
Network Locale: US
Emergency Response Location: 6
lpcor type: remote
lpcor (incoming): local_sccp_phone_1     (outgoing): local_sccp_phone_1
```

The following table describes significant fields in this output.

Table 53: show telephony-service ephone Field Descriptions

Field	Description
ephone-template	Identifier for the ephone template.
softkey hold	Soft keys displayed during the hold call stage.
softkey idle	Soft keys displayed during the call-idle call stage.
softkey seized	Soft keys displayed during the call-seized call stage.
softkey alerting	Soft keys displayed during the call-alerting call stage.
softkey connected	Soft keys displayed during the call-connected call stage.
conference drop-mode	When conferences are dropped: • creator: Conference terminates when the creator hangs up. • local: Conference terminates when the last local party in the conference hangs up or drops out of the conference. • never: Conference is not dropped, even if the creator hangs up, if three parties remain in the conference.
conference add-mode	Who can add parties to a conference: • creator: Only the creator can add parties. • all: Any party can add other parties if the creator remains in the conference.
conference admin	Ad hoc and meet-me hardware conference administrator. The administrator can: • Dial in to any conference directly through the conference number. • Use the ConfList soft key to list conference parties. • Remove any party from any conference.
Always send media packets to this router	Always send media packets to this Cisco Unified CME router, which acts as a proxy and forwards the packets to the destination, instead of sending them directly to the destination IP phone.
Preferred codec	Codec to use when initiating a call.

Field	Description
button-layout	Type of IP phone and number of fixed line or feature set. • 1: Button 24=Menu. Button 23=Headset. • 2: Button 24=Menu. Button 23=Headset. Button 22=Directories. Button 21=Messages.
User Locale	Locale that is associated with the phone user interface. The user locale identifies a set of detailed information, including language and font, to support users.
Network Locale	Locale that is associated with the phone. The network locale contains a definition of the tones and cadences that are used by the phones and gateways in the device pool in a specific geographic area.
Emergency response location	Identification of the ERL defined with the **emergency response location** command.
lpcor (incoming)	Setting of the **lpcor incoming** command.
lpcor (outgoing)	Setting of the **lpcor outgoing** command.
lpcor type	Setting of the **lpcor type** command.

Related Commands

Command	Description
ephone-template	Creates an ephone template.

show telephony-service fac

To display current feature access codes (FACs), use the **show telephony-service fac** command in privileged EXEC mode.

show telephony-service fac

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Phone users dial FACs to access phone features. The set of standard FACs must be enabled using the **fac standard** before phone users can use them. Individual FACs can be changed using the **fac custom** command.

Examples

The following example displays the set of standard FACs:

```
Router# show telephony-service fac
telephony-service fac standard
 callfwd all **1
 callfwd cancel **2
 pickup local **3
 pickup group **4
 pickup direct **5
 park **6
 dnd **7
 redial **8
 voicemail **9
 ephone-hunt join *3
 ephone-hunt cancel #3
```

Related Commands

Command	Description
fac	Enables standard FACs or creates a custom FAC.

show telephony-service security-info

To display the security-related information that is configured under telephony-service, use the **show telephony-service security-info** command in privileged EXEC configuration mode.

show telephony-service security-info

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example displays security information that was configured under telephony-service.

```
Router# show telephony-service security-info
Skinny Server Trustpoint for TLS: cisco1
TFTP Credentials Trustpoint: cisco1
Server Security Mode: Secure
Global Device Security Mode: Authenticated
```

show telephony-service tftp-bindings

To display the current configuration files accessible to IP phones, use the **show telephony-service tftp-bindings** command in user EXEC or privileged EXEC mode.

show telephony-service tftp-bindings

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Use this command with Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version.

This command provides a list of configuration files that are accessible to IP phones using TFTP, including the dictionary, language, and tone configuration files that are associated with the ISO-3166 codes that have been selected using the **user-locale** and **network-locale** commands.

Examples

The following is sample output from the **show telephony-service tftp-bindings** when the ISO-3166 code for Germany has been selected for both language and tones:

```
Router(config)# show telephony-service tftp-bindings
tftp-server system:/its/SEPDEFAULT.cnf
tftp-server system:/its/SEPDEFAULT.cnf alias SEPDefault.cnf
tftp-server system:/its/XMLDefault.cnf.xml alias XMLDefault.cnf.xml
tftp-server system:/its/ATADefault.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEP00036B54BB15.cnf.xml
tftp-server system:/its/germany/7960-font.xml alias German_Germany/7960-font.xml
tftp-server system:/its/germany/7960-dictionary.xml alias German_Germany/7960-dictionary.xml
tftp-server system:/its/germany/7960-kate.xml alias German_Germany/7960-kate.xml
tftp-server system:/its/germany/SCCP-dictionary.xml alias German_Germany/SCCP-dictionary.xml
tftp-server system:/its/germany/7960-tones.xml alias Germany/7960-tones.xml
```

Related Commands

Command	Description
network-locale	Sets the definition of the tones and cadences on the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G for a specific geographic area.
user-locale	Sets language for displays on the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G.

show telephony-service voice-port

To display configurations of virtual voice ports in a Cisco CallManager Express (Cisco CME) system, use the **show telephony-service voice-port** command in user EXEC or privileged EXEC mode.

show telephony-service voice-port

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	1.0	This command was introduced.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

This command displays virtual voice-port configurations for a Cisco CME system. Each ephone-dn corresponds to a virtual voice port. For example, the ephone-dn with dn-tag 7 corresponds to virtual voice port 50/0/7. The virtual voice port provides the telephone line associated with the Cisco IP phone extension (ephone-dn).

Examples

The following is sample output from this command:

```
Router# show telephony-service voice-port
voice-port 50/0/1
 station-id number 5001
!
voice-port 50/0/2
 station-id number 5002
 timeout ringing 8
!
voice-port 50/0/3
 station-id number 5003
!
voice-port 50/0/4
 station-id number 5004
!
```

The following table describes significant fields in this output, in alphabetical order.

Table 54: show telephony-service voice-port Field Descriptions

Field	Description
station-id number	Phone number used for caller ID purposes for calls made from this voice port.
timeout ringing	Maximum amount of time that a phone is allowed to ring before the call is disconnected.
voice-port	Virtual voice port.

Related Commands

Command	Description
show telephony-service all	Displays the detailed configuration of all the Cisco IP phones.
show telephony-service dial-peer	Displays dial-peer information for extensions in a Cisco CME system.
show telephony-service ephone-dn	Displays information for extensions (ephone-dns) in a Cisco CME system.

show voice emergency

To display the IP address, subnet mask, and ELIN for each emergency response location, use the **show voice emergency** command in user EXEC or privileged EXEC mode.

show voice emergency

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to display the IP address, subnet mask, and ELIN for each emergency response location.

Examples

The following example shows sample output which includes IP mask and ELIN information for each ERL:

```
EEMERGENCY RESPONSE LOCATIONS
ERL                 | ELIN 1     | ELIN2      | SUBNET 1        | SUBNET 2
1                   | 6045550101 |            | 10.0.0.0        | 255.0.0.0
2                   | 6045550102 | 6045550106 | 192.168.0.0     | 255.255.0.0
3                   |            | 6045550107 | 172.16.0.0      | 255.255.0.0
4                   | 6045550103 |            | 192.168.0.0     | 255.255.0.0
5                   | 6045550105 |            | 209.165.200.224 | 255.0.0.0
6 6045550198        |            | 6045550109 | 209.165.201.0   | 255.255.255.224
```

Related Commands

Command	Description
voice emergency response location	Creates a tag for identifying an ERL for E911 services.

show voice emergency addresses

To display address information for each emergency response location, use the **show emergency addresses** command in user EXEC or privileged EXEC mode.

show voice emergency addresses

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command displays the physical address of each emergency response location.

Examples

The following example shows a sample output which includes physical address information for the ERL:

```
Router# show voice emergency addresses
3850 Zanker Rd, San Jose,604,5550101
225 W Tasman Dr, San Jose,604,5550102
275 W Tasman Dr, San Jose,604,5550103
518 Bellew Dr,Milpitas,604,5550104
400 Tasman Dr,San Jose,604,5550105
    3675 Cisco Way,San Jose,604,5550106
```

Related Commands

Command	Description
address	Specifies a comma separated text entry (up to 250 characters) of an ERL's civic address.
show voice emergency all	Displays all emergency response location information.
voice emergency response location	Creates a tag for identifying an ERL for E911 services.

show voice emergency all

To display all emergency response location information, use the **show voice emergency all** command in user EXEC or privileged EXEC mode.

show voice emergency all

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to display all information configured for each emergency response location.

Examples

The following example shows a sample output, displaying all ERL-related information for ERL 1 and 3.

```
VOICE EMERGENCY RESPONSE SETTINGS
   Callback Number: 6045550103
   Emergency Line ID Number: 6045550155
   Expiry: 2 minutes
   Logging Enabled
EMERGENCY RESPONSE LOCATION 1
   Name: Cisco Systems 1
   Address: 3850 Zanker Rd, San Jose,elin.1.3,elin.4.10
   IP Address 1: 209.165.200.226 IP mask 1: 255.255.255.254
   IP Address 2: 209.165.202.129 IP mask 2: 255.255.0.0
   Emergency Line ID 1: 6045550180
   Emergency Line ID 2:
   Last Caller:  6045550188 [Jan 30 2007 16:05.52 PM]
   Next ELIN For Emergency Call: 6045550166
EMERGENCY RESPONSE LOCATION 3
   Name: Cisco Systems 3
   Address: 225 W Tasman Dr, San Jose,elin.1.3,elin.4.10
   IP Address 1: 209.165.202.133 IP mask 1: 255.255.0.0
   IP Address 2: 209.165.202.130 IP mask 2: 255.0.0.0
   Emergency Line ID 1:
   Emergency Line ID 2: 6045550150
   Last Caller:
   Next ELIN For Emergency Call: 6045550151
```

Related Commands

Command	Description
address	Specifies a comma separated text entry (up to 250 characters) of an ERL's civic address.
elin	Specifies a PSTN number that will replace the caller's extension.
name	Specifies a string (up to 32-characters) used internally to identify or describe the emergency response location.
subnet	Defines which IP phones are part of this ERL.
voice emergency response location	Creates a tag for identifying an ERL for the E911 services.

show voice emergency callers

To display a list of 911 calls made over the last three hours, use the **show emergency callers** command in privileged EXEC mode.

show voice emergency callers

Syntax Description

This command has no arguments or keywords.

Command Default

No list of 911 calls is displayed.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added to Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to display a list of all 911 calls made in the past three hours. The list shows the originating number, the ELIN used, and the time the call was placed.

Examples

The following example shows sample output, which includes the originating number, the ELIN used, and the time the call was placed:

```
router# show voice emergency callers
EMERGENCY CALLS CALL BACK TABLE
ELIN                      | CALLER                      | TIME
6045550181                | 8155550151                 | Oct 12 2006 04:05:21
6045550182                | 8155550152                 | Oct 12 2006 04:05:21
```

Related Commands

Command	Description
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.

show voice emergency zone

To display each emergency response zone's list of locations in priority order, use the **show voice emergency zone** command in user EXEC or privileged EXEC mode.

show voice emergency zone

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>)
Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to display a list of the locations, in priority order, of all configured emergency response zones.

Examples

The following example shows a sample output which displays the ERL locations for emergency response zones 90 and 100.

```
EMERGENCY RESPONSE ZONES
 zone 90
    location 4
    location 5
    location 6
    location 7
    location 2147483647
 zone 100
    location 1 priority 1
    location 2 priority 2
    location 3 priority 3
```

Related Commands

Command	Description
location	Identifies locations within an emergency response zone.
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.
voice emergency response zone	Creates an emergency response zone within which ERLs can be grouped.

show voice fac statistics

To display the FAC failure statistics collected by the system, use the **show voice fac statistics** command in privileged EXEC mode.

show voice fac statistics

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to display the forced athentication code (FAC) success or failure statistics collected by the system.

Examples

The following is sample output from this command displaying all statistical information:

```
Router# show voice fac statistics
Voice FAC statistics for failure calls:
  Total basic calls:            5
  Total forward calls:          1
```

Related Commands

Command	Description
show call active voice	Displays call information for voice calls that are in progress.
show call history voice	Displays the call history table for voice calls.

show voice hunt-group

To display configuration information associated with one or all voice hunt groups in a Cisco Unified Communications Manager Express (Cisco Unified CME) system, use the **show voice hunt-group** command in privileged EXEC mode.

show voice hunt-group *hunt-group-tag* [**brief**] {**longest-idle** | **parallel** | **peer** | **sequential**}

Syntax Description

hunt-group-tag	(Optional) Unique sequence number that identifies the voice hunt group. Range is 1 to 100.
brief	(Optional) Displays brief information on all voice hunt groups in a Cisco CME system.
longest-idle	(Optional) Displays summary of longest-idle voice hunt groups.
parallel	(Optional) Displays summary of parallel voice hunt groups.
peer	(Optional) Displays summary of peer voice hunt groups.
sequential	(Optional) Displays summary of sequential voice hunt groups.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(24)T	This command was introduced in a release earlier than Cisco IOS Release 12.4(24)T.
15.2(2)T	This command was modified to add stat collect as a field.

Usage Guidelines

Use the **show voice hunt-group** command to get information about voice hunt group configuration on the gateway as an alternative to the **show running-confi**g command.

Use the **show voice hunt-group** and **show voice hunt-group brief** commands to display hunt group configuration information for all voice hunt groups in a Cisco Unified CME system. Use **show voice hunt-group** *hunt-group-tag* to display data on a specific hunt-tag configuration created by the **voice hunt-group** command. Use the **longest-idle**, **parallel**, **peer**, or **sequential** keywords to display data on a specific type of voice hunt group configuration created by the **voice hunt-group** command.

Examples

The following is a sample output from the **show voice hunt-group** command, displaying all voice hunt groups configured on the router:

```
Router# show voice hunt-group
Group 1
    type: longest-idle
    preference: 0
    preference (sec): 0
    timeout: 0
    final_number: 1
Group 34
    type: parallel
    pilot number: 3, peer-tag 2147483647
    secondary number: 4, peer-tag 2147483646
```

```
    preference: 0
    preference (sec): 0
    timeout: 0
    final_number:
```

The following is a sample output from the **show voice hunt-group** command, displaying the configuration for all the configured voice hunt groups:

```
Router# show voice hunt-group
Group 5
 type: parallel
 pilot number: 1234, peer-tag 1234
 list of numbers:
  MEMBER        USED_BY      STATE     LOGIN/LOGOUT
  =======       =======      ========  ============
  9498889994    9498889994   DOWN      Logout
  9498889993    9498889994   UP        Login
  *             -            -         -
 secondary number: 5678, peer-tag 5678
 list preference: 5
 preference (sec): 8
 timeout: 180
 final_number: 4444
Group 8
 type: longest-idle
 pilot number: 6666, peer-tag 6666
 list of numbers:
    MEMBER        USED_BY      STATE     LOGIN/LOGOUT
    =======       =======      ========  ============
    5106575902    5106575902   UP        Login
    4088531111    4088531111   UP        Login
    4083911375    4083911375   DOWN      Login
    4089306067    4089306067   DOWN      Logout
    8869395033    8869395033   DOWN      Logout
    88686619633   88686619633  DOWN      -
 preference: 0
 preference (sec): 0
 timeout: 180
 final_number:
 hops: 6
 phone-display: Yes
Group 10
 type: longest-idle
 pilot number: 7777777, peer-tag 7777777
 secondary number: 88888888, peer-tag 88888888
 list of numbers:
    MEMBER      USED_BY     STATE     LOGIN/LOGOUT
    =======     =======     ========  ============
    7654321     7654321     DOWN      Logout
    87654321    87654321    UP        Login
    987654321   987654321   UP        Logout
 preference: 0
 preference (sec): 0
 timeout: 180
 final_number:
 hops: 3
 phone-display: No
Group 15
 type: peer
 pilot number: 56789, peer-tag 56789
 list of numbers:
    MEMBER      USED_BY     STATE     LOGIN/LOGOUT
    =======     =======     ========  ============
```

```
    87654321    87654321    DOWN      Login
    9876        9876        UP        Logout
    87654       87654       DOWN      -
 preference: 0
 preference (sec): 0
timeout: 180
 final_number:
 hops: 3
 phone-display: Yes
```

The following is a sample output from the **show voice hunt-group** command, displaying information for a particular voice hunt group as specified by the *hunt-group-tag* number:

```
Router# show voice hunt-group 5
Group 5
 type: parallel
 pilot number: 1234, peer-tag 1234
 secondary number: 5678, peer-tag 5678
 list of numbers:
    MEMBER         USED_BY       STATE     LOGIN/LOGOUT
    =======        =======       ========  ============
    9498889994     9498889994    UP        Logout
    9498889993     9498889993    DOWN      Login
    *              -             -         -
 preference: 5
 preference (sec): 8
 timeout: 20
 final_number: 4444
```

The following is a sample output from the **show voice hunt-group** command, displaying information about all the voice hunt groups of a particular type:

```
Router# show voice hunt-group longest-idle

Group 8
 type: longest-idle
 pilot number: 6666, peer-tag 6666
 list of numbers:
    MEMBER         USED_BY       STATE     LOGIN/LOGOUT
    =======        =======       ========  ============
    5106575902     5106575902    UP        Logout
    4088531111     4088531111    UP        Login
    4083911375     4083911375    DOWN      -
    4089306067     4089306067    UP        Logout
    8869395033     8869395033    -         -
    88686619633    88686619633   UP        Login
 preference: 0
 preference (sec): 0
 timeout: 180
 final_number:
 hops: 6
 phone-display: Yes
Group 10
 type: longest-idle
 pilot number: 7777777, peer-tag 7777777
 secondary number: 88888888, peer-tag 88888888
 list of numbers:
    MEMBER    USED_BY    STATE     LOGIN/LOGOUT
    =======   =======    ========  ============
    7654321   7654321    UP        Logout
    87654321  87654321   UP        Login
    987654321 987654321  DOWN      Logout
 preference: 0
```

```
 preference (sec): 0
 timeout: 180
 final_number:
 hops: 3
 phone-display: No
```

The following is a sample output from the **show voice hunt-group** command with the keyword **brief**:

```
Router# show voice hunt-group brief
TAG TYPE PILOT     LIST
=== ==== ========  =====================================================
5   PAR  1234      9498889-, 9498889-
         5678      9498889-, 9498889-
8   LON  6666      5106575-, 4088531-, 4083911-, 4089306-, 8869395-,.....
10  LON  7777777   7654321, 8765432-, 9876543-
         8888888-  7654321, 8765432-, 9876543-
15  PER  56789     8765432-, 9876, 87654
```

The following is a sample output from the **show voice hunt-group** command, indicating that call statistics is being collected:

```
Router# show voice hunt-group 1
Group 1
    type: parallel
    pilot number: 5000, peer-tag 2147483647
    list of numbers:
    MEMBER   USED_BY   STATE     LOGIN/LOGOUT
    =======  =======   ========  ============
    5001     5001      UP        Logout
    5002     5002      UP        Login
    5011     5011      DOWN      -
    5012     5012      UP        Logout
    preference: 0
    preference (sec): 0
    timeout: 12
    final_number: 5012
    stat collect: yes
    phone-display: Yes
```

The following is a sample output from the **show voice hunt-group** command when there is no voice hunt group configured:

```
Router# show voice hunt-group
 no voice hunt-groups configured
Router# show voice hunt-group brief
 no voice hunt-groups configured
Router# show voice hunt-group longest-idle

 no voice hunt-groups configured
Router#
```

The following table describes the significant fields shown in the output.

Table 55: show voice hunt-group Field Descriptions

Field	Description
Group	Tag number of voice hunt group.

Field	Description
type	Type of voice hunt group. The available voice hunt group types are: longest-idle, parallel, peer and sequential.
pilot number	Number that callers dial to reach the specified voice hunt group.
secondary-number	Alternate number for the specified voice hunt group.
list of numbers	Numbers of the extensions configured in the **voice hunt-group** command‘s hunt-tag identifier.
preference	Preference order for the extension or telephone number associated with a dial peer. Range is 0 to 8. Default is 0.
preference (sec)	Preference order for the secondary pilot number. Range is from 0 to 10, where 0 is the highest preference and 10 is the lowest preference. Default is 9.
timeout	Number of seconds after which a call that is not answered at one number is redirected to the next number in the hunt group list.
final_number	Last number in the voice hunt group, after which a call is no longer redirected.
hops	Number of hops before a call proceeds to the final number.
stat collect	Yes indicates that call statistics are being collected for a voice hunt group.
phone-display	Displays the hunt group information on My Phone Apps service button.
hlog-block	Blocks the hlog functionality of voice hunt group on the phone.
peer-tag	Peer hunting tag.

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next directory number in a peer voice hunt group list before proceeding to the final directory number.
list (voice hunt-group)	Defines the directory numbers that participate in a directory number hunt group.
pilot **(voice hunt-group)**	Defines the voice dn that callers dial to reach a Cisco Unified Communications Manager Express (Cisco Unified CME) voice hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last directory number in the hunt group.
voice hunt-group	Configures voice hunt groups and the associated parameters.

show voice hunt-group statistics

To display call statistics from voice hunt groups, use the **show voice hunt-group statistics** command in privileged EXEC mode.

show voice hunt-group *group-id* **statistics** {**last** *hours* **hours** | **start** *day time* [**to** *day time*]}

Syntax Description

group-id	Identifier for the voice hunt group. Range: 1 to 100.
last	Displays the latest call statistics for a voice hunt group for a specified number of hours, counting backward from the current hour. Range: 1 to 167.
hours **hours**	Number of hours that the call statistics are displayed.
start	Defines the start of the period for which the call statistics are displayed. Default duration is one hour.
day	Abbreviated day of the week. The following abbreviations are valid: sun, mon, tue, wed, thu, fri, sat.
time	Hour of the day. Range: 0 to 23.
to	(Optional) Defines the time the display of the call statistics ends.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **show voice hunt-group statistics** command to display the average and longest times for a voice hunt group to answer a call, make a call, or put a call on hold. The command can also display the number of answered and abandoned calls, the number of calls forwarded to or answered by voice mail, and the number of error calls.

The output is dependent on call activity. If there is no activity, no data is displayed.

If your Cisco Unified CME system is configured with the basic automatic call distribution (B-ACD) and auto-attendant service, you can enable the collection of call statistics for every voice hunt group with the **voice hunt-group statistics collect** command. Additional data is displayed for all agents combined and for individual agents.

Note On the day that daylight saving time adjusts the time back by one hour at 2 a.m. each year, the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

For remote Cisco Unified SCCP IP phones in voice hunt groups, the hold and resume statistics are not updated.

Examples

The following is a sample output from the **show voice hunt-group statistics** command. The output includes direct calls to a voice hunt group number and calls from queue/B-ACD.

```
Router# show voice hunt-group 1 statistics last 1 h
Wed 04:00 - 05:00
 Max Agents: 3
 Min Agents: 3
 Total Calls: 9
 Answered Calls: 7
 Abandoned Calls: 2
 Average Time to Answer (secs): 6
 Longest Time to Answer (secs): 13
 Average Time in Call (secs): 75
 Longest Time in Call (secs): 161
 Average Time before Abandon (secs): 8
 Calls on Hold: 2
 Average Time in Hold (secs): 16
 Longest Time in Hold (secs): 21
 Per agent statistics:
  Agent: 5012
   From Direct Call:
    Total Calls Answered: 3
    Average Time in Call (secs): 70
    Longest Time in Call (secs): 150
    Totals Calls on Hold: 1
    Average Hold Time (secs): 21
    Longest Hold Time (secs): 21
   From Queue:
    Total Calls Answered: 3
    Average Time in Call (secs): 55
    Longest Time in Call (secs): 78
    Total Calls on Hold: 2
    Average Hold Time (secs): 19
    Longest Hold Time (secs): 26
  Agent: 5013
   From Direct Call:
    Total Calls Answered: 3
    Average Time in Call (secs): 51
    Longest Time in Call (secs): 118
    Totals Calls on Hold: 1
    Average Hold Time (secs): 11
    Longest Hold Time (secs): 11
   From Queue:
    Total Calls Answered: 1
    Average Time in Call (secs): 4
    Longest Time in Call (secs): 4
  Agent: 5014
   From Direct Call:
    Total Calls Answered: 1
    Average Time in Call (secs): 161
    Longest Time in Call (secs): 161
   From Queue:
    Total Calls Answered: 1
    Average Time in Call (secs): 658
    Longest Time in Call (secs): 658
 Queue related statistics:
  Total calls presented to the queue: 5
  Calls handoff to IOS: 5
  Number of calls in the queue: 0
  Average time to handoff (secs): 2
  Longest time to handoff (secs): 3
  Number of abandoned calls: 0
```

```
  Average time before abandon (secs): 0
  Calls forwarded to voice mail: 0
  Calls answered by voice mail: 0
  Number of error calls: 0
```

The following is a sample output from the **show voice hunt-group statistics** command. The output focuses on queue-related statistics.

```
Queue related statistics:
      Total calls presented to the queue: 8
      Calls handoff to IOS: 3
      Number of calls in the queue: 1
      Average time to handoff (secs): 10
      Longest time to handoff (secs): 15
      Number of abandoned calls: 4
      Average time before abandon (secs): 7
      Calls forwarded to voice mail: 0
      Calls answered by voice mail: 0
      Number of error calls: 0
```

The following is a sample output from the **show voice hunt-group statistics** command. The output shows that no call statistics were collected from voice hunt group 1 from 2:00 to 4:00 on a Monday.

```
Router# show voice hunt-group 1 stat start Mon 2 to Mon 4
Mon 02:00 - 03:00
    No info
Mon 03:00 - 04:00
    No info
Mon 04:00 - 05:00
    No info
```

The following table describes the significant fields shown in the display.

Table 56: show voice hunt-group statistics Field Descriptions

Field	Description
Abandoned calls	Total number of calls abandoned by hunt group agents. This does not include calls going to the final number.
Answered call	Total number of calls answered by hunt group agents.
Average time in call (secs)	Average length of time that unanswered calls waited before going to an agent.
Average time to answer (secs)	Average length of time that all calls to Cisco Unified CME B-ACD waited before being answered.
Average time in hold (secs)	Average length of time that calls were kept on hold for all agents.
Average hold time (secs)	Average length of time that calls waited on hold for this agent.
Average time to handoff (secs)	Average length of time before a call was handed off to IOS
Calls on hold	Total number of calls that were placed on hold.
Calls handoff to IOS	Total number of calls handed off to IOS.

Field	Description
Calls answered by voice mail	Total number of calls to Cisco Unified CME B-ACD that were answered by voice mail.
Calls forwarded to voice mail	Total number of calls to Cisco Unified CME B-ACD that were forwarded to voice mail.
Longest time to answer (secs)	Longest length of time before calls to Cisco Unified CME B-ACD were answered.
Longest time in call (secs)	Longest length of time that all calls to Cisco Unified CME B-ACD that went to an agent waited in a call queue.
Longest time in hold (secs)	Longest length of time that a call spent between being placed on hold and being picked up by agents.
Longest hold time (secs)	Longest length of time that a call to this agent was spent between being placed on hold and being picked up.
Longest time to handoff (secs)	Longest length of time before a call was handed off to IOS.
Max agent	Maximum number of hunt group agents.
Min agent	Minimum number of hunt group agents.
Number of abandoned calls	Total number of calls to Cisco Unified CME B-ACD that hung up before being answered.
Number of calls in the queue	Total number of calls in the queue.
Number of error calls	Total number of error calls.
Total calls	Total number of direct calls made to the hunt group.
Total calls answered	Total number of calls to Cisco Unified CME B-ACD that were answered by an agent.
Total calls on hold	Total number of calls that were placed on hold for this agent.
Total calls presented to the queue	Total number of calls made to Cisco Unified CME B-ACD.

Note From Cisco Unified CME Release 10.5 onwards, abandoned calls will not include the calls going to the final number. However, the total calls includes calls going to the final number. Use the formula " **Final Calls= Total Calls - Answered Calls - Abandoned Calls**" , to calculate the calls going to the final number.

Related Commands

Command	Description
voice hunt-group statistics collect	Enables the collection of call statistics for voice hunt groups.

show voice register all

To display all Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified Communications Manager Express (Cisco Unified CME) configurations and register information, use the **show voice register all** command in privileged EXEC mode.

show voice register all

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.
15.0(1)XA	Cisco SIP SRST 8.0	This command was modified to display the signaling transport protocol.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1	The output display of this command was modified.
15.2(4)M	Cisco Unified CME 9.1	This command was modified to include Key Expansion Module (KEM) data in the output display.

Usage Guidelines

KEM data are displayed for Cisco Unified CME only. Cisco Unified SRST is unable to gather all the configuration details about KEMs from Cisco Unified CM.

Examples

Cisco Unified SIP SRST

The following is a sample output of the **show voice register all** command:

```
Router# show voice register all
VOICE REGISTER GLOBAL
=====================
CONFIG [Version=8.1]
========================
  Version 8.1
  Mode is srst
  Max-pool is 10
  Max-dn is 10
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  timeout interdigit 10
  network-locale[0] US    (This is the default network locale for this box)
```

```
  network-locale[1] US
  network-locale[2] US
  network-locale[3] US
  network-locale[4] US
  user-locale[0] US    (This is the default user locale for this box)
  user-locale[1] US
  user-locale[2] US
  user-locale[3] US
  user-locale[4] US   Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
VOICE REGISTER DN
=================
Dn Tag 1
Config:
  Number is 45111
  Preference is 0
  Huntstop is disabled
  Pool 1    has this DN configured for line 1
Dn Tag 2
Config:
  Number is 45112
  Preference is 0
  Huntstop is disabled
  Pool 2    has this DN configured for line 1
Dn Tag 3
Config:
  Number is 45113
  Preference is 0
  Huntstop is disabled
  Pool 3    has this DN configured for line 1, 2
Dn Tag 4
Config:
Dn Tag 7
Config:
  Number is 451110
  Preference is 0
  Huntstop is disabled
  Pool 1    has this DN configured for line 4
Dn Tag 8
Config:
  Pool 1    has this DN configured for line 3
VOICE REGISTER POOL
===================
 Pool Tag 1
Config:
  Mac address is 001B.535C.D410
  Number list 1 : DN 1
  Number list 3 : DN 8
  Number list 4 : DN 7
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
```

```
  kpml signal is disabled
  Lpcor Type is none
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 2
Config:
  Mac address is 0015.C68E.6D13
  Number list 1 : DN 2
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is disabled
  Lpcor Type is none
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 3
Config:
  Mac address is 0021.5553.8998
  Number list 1 : DN 3
  Number list 2 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is enabled
  Lpcor Type is none
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
```

```
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
```

Cisco Unified CME

The following is a sample output of the **show voice register all** command:

```
Router# show voice register all
1) show voice register all
VOICE REGISTER GLOBAL
=====================
CONFIG [Version=8.1]
========================
  Version 8.1
  Mode is cme
  Max-pool is 10
  Max-dn is 10
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  Source-address is 8.3.3.5 port 5060
  Time-format is 12
  Date-format is M/D/Y
  Time-zone is 5
  Hold-alert is disabled
  Mwi stutter is disabled
  Mwi registration for full E.164 is disabled
  Forwarding local is enabled
  Privacy is enabled
  Privacy-on-hold is disabled
  Dst auto adjust is enabled
    start at Apr week 1 day Sun time 02:00
    stop  at Oct week 8 day Sun time 02:00
  Max redirect number is 5
  IP QoS DSCP:
    ef (the MS 6 bits, 46, in ToS, 0xB8) for media
    cs3 (the MS 6 bits, 24, in ToS, 0x60) for signal
    af41 (the MS 6 bits, 34, in ToS, 0x88) for video
    default (the MS 6 bits, 0, in ToS, 0x0) for service
  Telnet Level: 0
  Tftp path is flash:
  Generate text file is disabled
  Tftp files are created, current syncinfo 0001140473454008
  OS79XX.TXT is not created
  timeout interdigit 10
  network-locale[0] US    (This is the default network locale for this box)
  network-locale[1] US
  network-locale[2] US
  network-locale[3] US
  network-locale[4] US
  user-locale[0] US    (This is the default user locale for this box)
  user-locale[1] US
```

```
  user-locale[2] US
  user-locale[3] US
  user-locale[4] US   Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
VOICE REGISTER DN
=================
Dn Tag 1
Config:
  Number is 45111
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  Pool 1    has this DN configured for line 1
Dn Tag 2
Config:
  Number is 45112
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua noan 999 timeout 8
  after-hour exempt
  Pool 2    has this DN configured for line 1
  Pool 7    has this DN configured for line 1
Dn Tag 3
Config:
  Number is 45113
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua all 87687
  Pool 3    has this DN configured for line 1, 2
Dn Tag 4
Config:
  Auto answer is disabled
Dn Tag 7
Config:
  Number is 451110
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  after-hour exempt
  Pool 1    has this DN configured for line 4
Dn Tag 8
Config:
  Auto answer is disabled
  call-forward b2bua all 678
  after-hour exempt
  Pool 1    has this DN configured for line 3
VOICE REGISTER TEMPLATE
=======================
Temp Tag 1
```

```
Config:
  Attended Transfer is enabled
  Blind Transfer is enabled
  Semi-attended Transfer is enabled
  Conference is enabled
  Caller-ID block is disabled
  DnD control is enabled
  Anonymous call block is disabled
  Dialplan Tag is 1
  softkey connected  Confrn
  Lpcor type none
  Pool 4 has this template configured
VOICE REGISTER DIALPLAN
=======================
Dialplan Tag 1
Config:
  Type is 7905-7912
  Template 1 has this dialplan configured
  Pool 4 has this dialplan configured
VOICE REGISTER POOL
===================
 Pool Tag 1
Config:
  Mac address is 001B.535C.D410
  Type is 7960
  Number list 1 : DN 1
  Number list 3 : DN 8
  Number list 4 : DN 7
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  call-forward phone all is 4566
  call-forward b2bua all 4555
  keep-conference is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 2
Config:
  Mac address is 0015.C68E.6D13
  Type is 7960
```

```
  Number list 1 : DN 2
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  call-forward phone noan is 9886, timeout 98
  keep-conference is enabled
  username pool2 password lab
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0

  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 3
Config:
  Mac address is 0021.5553.8998
  Type is 7975
  Number list 1 : DN 3
  Number list 2 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is enabled
  Busy trigger per button value is 0
  call-forward phone all is 45112
  call-forward b2bua all 45111
  after-hour exempt
  keep-conference is enabled
  kpml signal is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
```

```
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 4
Config:
  Mac address is 8989.9867.8769
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  keep-conference is enabled
  template is 1
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 7
Config:
  Mac address is 0018.BAC8.D2B1
  Number list 1 : DN 2
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  keep-conference is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
```

```
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
```

The following is an example of a partial output of the **show voice register all** command, showing KEM data with the phone type information:

```
Router# show voice register all
Pool Tag 5
Config:
  Mac address is B4A4.E328.4698
  Type is 9971 addon 1 CKEM
  Number list 1 : DN 2
  Number list 2 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Video is enabled
  Camera is enabled
  Busy trigger per button value is 0
  keep-conference is enabled
  registration expires timer max is 200 and min is 60
  kpml signal is enabled
  Lpcor Type is none
```

The following is a sample output of the **show voice register all** command, showing the three KEMs configured with phone type 9971:

```
Router# show voice register all
Pool Tag 4
Config:
  Mac address is B4A4.E328.4698
  Type is 9971 addon 1 CKEM 2 CKEM 3 CKEM
  Number list 1 : DN 4
  Number list 2 : DN 5
  Number list 3 : DN 9
```

The following table describes the significant fields shown in this output.

Table 57: show voice register all Field Descriptions

Field	Description
Pool Tag	Shows the assigned tag number of the current voice register pool.
Config	Shows the voice register pool.
Network address and Mask	Shows network address and mask information when the **id** command is configured.

Field	Description
Number list, Pattern, and Preference	Shows the **number** command configuration.
Proxy IP address	Shows the **proxy** command configuration.
Default preference	Shows the default preference value of this pool.
Incoming called number	Shows the **incoming called-number** command configuration.
Translate outgoing called tag	Shows the **translate-outgoing** command configuration.
Class of Restriction List Tag	Shows the COR tag.
Incoming corlist name	Shows the **cor** command configuration.
Application	Shows the **application** command configuration for this pool.
Dialpeers created	Lists all the dial peers created and their contents. Dial-peer contents differ for each application and are not described here.
Statistics	Shows the registration statistics for this pool.
Active registrations	Shows the current active registrations.
Total Registration Statistics	Shows the total registration statistics for this pool.
Registration requests	Shows the incoming registration requests.
Registration success	Shows the successful registrations.
Registration failed	Shows the failed registrations.
unRegister requests	Shows the incoming unregister/registration expire requests.
unRegister success	Reports the number of successful unregisters.
unRegister failed	Reports the number of failed unregisters.

Related Commands

Command	Description
application (voice register pool)	Selects the session-level application for the dial peer associated with an individual Cisco Unified SIP IP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.
cor (voice register pool)	Configures a class of restriction on the VoIP dial peers associated with directory numbers.
id (voice register pool)	Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.

Command	Description
incoming called-number (dial peer)	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.
number (voice register pool)	Indicates the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone.
proxy (voice register pool)	Autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco Unified SIP IP phone registers with a Cisco Unified SIP SRST gateway.
show sip-ua status registrar	Displays all the SIP endpoints currently registered with the contact address.
show voice register dial-peers	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified CME register event.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
translate-outgoing (voice register pool)	Allows an explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by any Cisco Unified IP phone user.

show voice register credential

To display configuration information associated with a credential file used for authorization, use the **show voice register credential** command in privileged EXEC mode.

show voice register credential

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Examples

The following is sample output from this command:

```
Router# show voice register credential
username: Jsmith, password: 1234abc, service: PRESENCE , file index 3
username: Ksample, password: xyz1234, service: PRESENCE , file index 3
username: Mmore, password: updwssc, service: PRESENCE , file index 3
username: Sstove, password: 12bms, service: PRESENCE , file index 3
username: Yjones, password: 357llvrus, service: PRESENCE , file index 5
username: Yjones2, password: 55rrtuv, service: PRESENCE OOD_REFER , file index 5
username: vtemp, password: 1234567, service: PRESENCE , file index 5
```

The table contains descriptions of fields shown in the output, listed in order of appearance.

Table 58: show voice register credential Field Descriptions

Field	Description
username	Username that is authorized.
password	Password that is authorized.
service	Type of service for which the credential file is used; presence or Out-of-dialog REFER (OOD-R).
file index	Identification number of the credential file defined with the **authenticate** command.

Related Commands

Command	Description
authenticate (voice register global)	Defines the authenticate mode for SIP phones in a Cisco Unified CME system.
credential load	Reloads a credential file into Flash memory.

Command	Description
show voice register all	Displays all Cisco Unified CME and Cisco Unified SIP SRST configurations and register information.

show voice register dial-peers

To display details of all dynamically created VoIP dial peers associated with the Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME) register event, use the **show voice register dial-peers** command in privileged EXEC mode.

show voice register dial-peers [**pool** **tag**]

Syntax Description

pool tag	Number of entries in attempted registrations table. Size range from 0 to 50.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1	This command was modified. Pool tag keyword-argument was added. Command output display was also modified to display dial-peers specific to a pool.

Usage Guidelines

Use this command to display the dial-peers associated with a pool. To display the dynamic dial-peers associated with a specific pool, use the pool keyword followed by the pool tag. When using the pool keyword, you must specify the pool tag.

Examples

Cisco Unified CME and Cisco Unified SIP SRST

The following is a sample output from this command displaying all dial-peers:

```
Router#show voice register dial-peers
Dial-peers for Pool 1
dial-peer voice 40001 voip
 destination-pattern 45111
 session target ipv4:8.3.3.111:5060
 session protocol sipv2
 call-fwd-all         4555
 after-hours-exempt   FALSE
dial-peer voice 40002 voip
  destination-pattern 45113
  session target ipv4:8.33.33.111:5060
  session protocol sipv2
  after-hours-exempt   FALSE
Dial-peers for Pool 2
```

```
 dial-peer voice 40003 voip
 destination-pattern 45112
 session target ipv4:8.33.33.112:5060
 session protocol sipv2
 call-fwd-noan-timeou 8
 call-fwd-noan        999
 after-hours-exempt   TRUE
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information related to pool 1:

```
Router# show voice register dial-peers pool 1
Dial-peers for Pool 1:
dial-peer voice 40004 voip
 destination-pattern 1000
 redirect ip2ip
 session target ipv4:9.13.18.40:19633
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
dial-peer voice 40001 voip
 destination-pattern 2000
 redirect ip2ip
 session target ipv4:9.13.18.40:19634
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
```

after-hours-exempt FALSE

Related Commands

Command	Description
show sip-ua status registrar	Displays all the SIP endpoints currently registered with the contact address.
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register dialplan

To display all configuration information for a specific SIP dial plan, use the **show voice register dialplan** command in privileged EXEC mode.

show voice register dialplan{*tag* | **all**}

Syntax Description

tag	Number that identifies the SIP dialplan. Range: 1 to 24.
all	(Optional) Displays all the dialplans defined in a system.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. All keyword was added. Pools and templates that have dialplan configured are also displayed in the output.

Usage Guidelines

Use this command to verify the configuration of SIP dial plans. You define a dial plan with the **voice register dialplan** command and assign it to a SIP phone with the **dialplan** command.

In Cisco Unified CME 8.1 and later, **show voice register dialplan** command also displays the pools and templates that have the dialplan configured. The pools which have the diaplan configured by virtue of inclusion of a template is also displayed as part of the pool list display. If a dialplan is configured under both template and pool, the dialplan under the pool takes precedence and the pool is displayed.

When used with the all keyword, the show voice register diaplan command displays configuration information for all the dialplans defined in a system.

Examples

The following is sample output from this command displaying information for dialplan 1:

```
Router# show voice register dialplan 1
Dialplan Tag 1
Config:
  Type is 7905-7912
  Template 1 has this dialplan configured
  Pool 4 has this dialplan configured
```

The following is a sample output from this command displaying information for all the dialplans configured in a system:

```
Router# show voice register dialplan all
Dialplan Tag 1
Config:
  Type is 7905-7912
```

```
  Pattern 1 is 9879, timeout is 0, user option is phone, button is default
  Pattern 24 is 908, timeout is 0, user option is phone, button is default
Dialplan Tag 2
Config:
  Type is 7940-7960-others
  Pattern 3 is 9845, timeout is 0, user option is phone, button is default
  Pattern 20 is 9098, timeout is 0, user option is phone, button is default
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 59: show voice register dialplan Field Descriptions

Field	Description
Config	List of configuration options defined for this SIP dial plan.
Dialplan Tag	Tag number of the requested SIP dial plan.
Pattern	Dial pattern defined for a SIP dial plan with the **pattern** command in voice register dialplan configuration mode.
Type	Phone type defined for a SIP dial plan with the **type** command.

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.
pattern (voice register dialplan)	Defines a dial pattern for a SIP dial plan.
show voice register all	Displays all Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
type (voice register dialplan)	Defines a phone type for a SIP dial plan.
voice register dialplan	Enters voice register dialplan configuration mode to define a dial plan for SIP phones.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voice register dn

To display all configuration information associated with a specific voice register dn, use the **show voice register dn** command in privileged EXEC mode.

show voice register dn{*tag* | **all**}

Syntax Description

tag	Tag number of the voice register dn for which to display information. Range is 1 to 750.
all	(Optional) Displays configuration information associated with all voice register dns defined in a system.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was introduced.
15.1(2)T	Cisco CME 8.1 and Cisco SIP SRST 8.1	This command was modified.The display output now shows pools that have DNs configured under them. All keyword was added to show configuration information for all voice register dns defined in system.

Usage Guidelines

In Cisco Unified CME 8.1 and Cisco Unified SIP SRST 8.1, the show voice register dn command displays the pools that have the DNs configured under them. When used with all keyword, the show voice register dn command displays configuration information for all the DNs defined in a system.

Examples

Cisco Unified SIP CME

The following is a sample output from this command:

```
Router# show voice register dn 1
Dn Tag 1
Config:
  Number is 11
  Preference is 10
  Huntstop is enabled
  Translation-profile incoming saaa
  Allow watch is enabled
  Pool 1    has this DN configured for line 1
```

Cisco Unified SIP SRST

The following is a sample output from this command:

```
Router# show voice register dn 2
Dn Tag 1
Config:
```

```
  Number is 11
  Preference is 10
  Huntstop is enabled
  Translation-profile incoming saaa
  Allow watch is enabled
  Pool 1    has this DN configured for line 1
```

Cisco Unified SIP SRST

The following is a sample output from this command displaying information for all the dns:

```
Dn Tag 1
Config:
  Number is 11
  Preference is 10
  Huntstop is enabled
  Translation-profile incoming saaa
  Allow watch is enabled
  Pool 1    has this DN configured for line 1
Dn Tag 2
Config:
  Number is 12
  Preference is 1
  Huntstop is enabled
  Allow watch is enabled
  Pool 2    has this DN configured for line 1, 2
```

Cisco Unified SIP CME

The following is a sample output from this command displaying information for all the dns:

```
Router# show voice register dn all
Dn Tag 1
Config:
  Number is 45111
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
Dn Tag 2
Config:
  Number is 45112
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua noan 999 timeout 8
  after-hour exempt
  Pool 2    has this DN configured for line 1
  Pool 7    has this DN configured for line 1
Dn Tag 3
Config:
  Number is 45113
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua all 87687
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua all 87687
```

```
  Pool 1    has this DN configured for line 1
  Pool 3    has this DN configured for line 1, 2
Dn Tag 4
Config:
  Auto answer is disabled
Dn Tag 7
Config:
  Number is 451110
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  after-hour exempt
  Pool 1    has this DN configured for line 4
Dn Tag 8
Config:
  Auto answer is disabled
  call-forward b2bua all 678
  after-hour exempt
  Pool 1    has this DN configured for line 3
```

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 60: show voice register dn Field Descriptions

Field	Description
Auto answer	Status of auto-answer feature defined with the **auto-answer** command.
Config	List of configuration options defined for this voice register dn.
Dn Tag	Tag number of the requested voice register dn.
Huntstop	Status of huntstop behavior defined with the **huntstop** command.
Number	Telephone or extension number set with the **number** command in voice register dn configuration mode.
Preference	Preference order set with the **preference** command in voice register dn configuration mode.

Related Commands

Command	Description
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register dn all	Displays information associated with all the dns configured in a system.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

show voice register global

To display all global configuration parameters associated with Cisco Unified SIP IP phones, use the **show voice register global** command in privileged EXEC mode.

show voice register global

Syntax Description

This command has no arguments or keywords.

Command Default

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
15.0(1)XA	Cisco SIP SRST 8.0	This command was modified to display the signaling transport protocol.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1	This command was modified to include global statistics in the output display.
15.2(2)T	Cisco Unified CME 9.0	This command was modified to include conference hardware in the output display.

Examples

Cisco Unified CME

The following is a sample output from the **show voice register global** command used in Cisco Unified CME:

```
Router# show voice register global
CONFIG [Version=8.1]
========================
  Version 8.1
  Mode is cme
  Max-pool is 10
  Max-dn is 10
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  Source-address is 8.3.3.5 port 5060
  Time-format is 12
  Date-format is M/D/Y
  Time-zone is 5
  Hold-alert is disabled
  Mwi stutter is disabled
  Mwi registration for full E.164 is disabled
  Forwarding local is enabled
  Privacy is enabled
  Privacy-on-hold is disabled
  Dst auto adjust is enabled
    start at Apr week 1 day Sun time 02:00
    stop  at Oct week 8 day Sun time 02:00
  Max redirect number is 5
  IP QoS DSCP:
```

```
    ef (the MS 6 bits, 46, in ToS, 0xB8) for media
    cs3 (the MS 6 bits, 24, in ToS, 0x60) for signal
    af41 (the MS 6 bits, 34, in ToS, 0x88) for video
    default (the MS 6 bits, 0, in ToS, 0x0) for service
  Telnet Level: 0
  Tftp path is flash:
  Generate text file is disabled
  Tftp files are created, current syncinfo 0001140473454008
  OS79XX.TXT is not created
  timeout interdigit 10
  network-locale[0] US    (This is the default network locale for this box)
  network-locale[1] US
  network-locale[2] US
  network-locale[3] US
  network-locale[4] US
  user-locale[0] US    (This is the default user locale for this box)
  user-locale[1] US
  user-locale[2] US
  user-locale[3] US
  user-locale[4] US   Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
```

The following is a sample output from the **show voice register global** command. The output shows that hardware conferencing is enabled.

```
Router# show voice register global
CONFIG [Version=8.7]
========================
  Version 8.7
  Mode is cme
  Max-pool is 50
  Max-dn is 100
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  Forced Authorization Code Refer is enabled
  Source-address is 1.5.40.20 port 5060
  Time-format is 12
  Date-format is M/D/Y
  Time-zone is 5
  Hold-alert is disabled
  Mwi stutter is disabled
  Mwi registration for full E.164 is disabled
  Forwarding local is enabled
  Video is enabled
  Camera is enabled
  Privacy is enabled
  Privacy-on-hold is disabled
  Conference hardware is enabled
  Dst auto adjust is enabled
```

```
start at Apr week 1 day Sun time 02:00
stop  at Oct week 8 day Sun time 02:00
```

Cisco Unified SIP SRST

The following is a sample output from the **show voice register global** command used in Cisco Unified SIP SRST:

```
Router# show voice register global
CONFIG [Version=8.1]
========================
  Version 8.1
  Mode is srst
  Max-pool is 10
  Max-dn is 10
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  timeout interdigit 10
  network-locale[0] US    (This is the default network locale for this box)
  network-locale[1] US
  network-locale[2] US
  network-locale[3] US
  network-locale[4] US
  user-locale[0] US    (This is the default user locale for this box)
  user-locale[1] US
  user-locale[2] US
  user-locale[3] US
  user-locale[4] US   Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
```

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 61: show voice register global Field Descriptions

Field	Description
Conference hardware	Shows whether the Cisco Unified SIP IP phone will perform local mixing on its own or request Cisco Unified CME to perform hardware conferencing using its DSP resource.
Date-format	Value of **date-format** command.
DST auto adjust	Setting of **dst auto-adjust** command.
Forwarding local	Setting of **forwarding local** command.

Field	Description
Generate text file	Setting of **text file** command.
Hold-alert	Setting of **hold-alert** command.
Load	Value of **load** command.
Max-dn	Reports the maximum number of SIP voice register directory numbers (DNs) supported by the Cisco Unified SIP CME or Cisco Unified SIP SRST router as configured with the **max-dn** command. The maximum possible number is platform-dependent.
Max-pool	Reports the maximum number of SIP voice register pools supported by the Cisco Unified SIP SRST or Cisco Unified CME router as configured with the **max-pool** command. The maximum possible number is platform-dependent.
Max redirect number	Maximum number of redirects set with the **max-redirect** command.
Mode	Reports the mode as configured with the **mode** command. Value can be either Cisco Unified CME or Cisco Unified SIP SRST.
MWI registration	Setting of **mwi** command.
MWI stutter	Setting of **mwi stutter** command.
Time-format	Value of **time-format** command.
Time-zone	Number of the timezone selected with the **timezone** command.
TFTP path	Directory location of provisioning files for Cisco Unified SIP IP phones that is specified with the **tftp-path** command.
Version	Reports the Cisco Unified SIP SRST or Cisco Unified CME version number.

Related Commands

Command	Description
show sip-ua status registrar	Displays all the Cisco Unified SIP IP phones currently registered with the contact address.
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register dial-peers	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified CME register event.
voice register global	Enters voice register global configuration mode to set global parameters for all supported Cisco Unified SIP IP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.

show voice register hfs

To display the HTTP File-Fetch Server (HFS) file bindings of firmware files accessible to Cisco Unified SIP IP phones, use the **show voice register hfs** command in privileged EXEC mode.

show voice register hfs

Syntax Description

This command has no arguments or keywords.

Command Default

None

Command Modes

Privileged EXEC

Command History

Release	Modification
15.2(1)T	This command was introduced.

Usage Guidelines

Use the **show voice register hfs** command with Cisco Unified CME 8.8 or a later version.

This command displays the bindings of firmware files that are accessible to Cisco Unified SIP IP phones using the HFS download service.

Examples

The following is a sample output from the **show voice register hfs** command:

```
Router(config)# show voice register hfs
Fetch Service Enabled = Y
   App enabled port = 6970
   Use default port = N
   Registered session-id = 19

Default home path = flash:/
   Ongoing fetches from home = 0

HTTP File Server Bindings
   No. of bindings = 11
   No. of url table entries = 9
   No. of alias table entries = 9
```

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profile files required for SIP phones.
hfs enable	Enables the HFS download service on an IP Phone in a Cisco Unified CME system.

show voice register pool

To display all configuration information associated with a specific voice register pool, use the **show voice register pool** command in privileged EXEC mode.

show voice register pool {*pool-tag* | **all**} [**brief**]

Syntax Description

pool-tag	Tag number of the voice register pool for which information is displayed. Range is 1 to 262. **Note** The maximum number of pools is version and platform dependent.
all	Displays the information of all the voice register pools.
brief	(Optional) Displays brief information of all voice register pools.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST	This command was introduced.
12.3(4)T	Cisco SIP SRST	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was modified to include emergency response location information in the output display.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified to include logical partitioning class of restriction (LPCOR) information in the output display.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. The **all** and **brief** keywords were added. Voice-class stun-usage information is displayed in the output.
15.2(2)T	Cisco Unified CME 9.0	This command was modified to include conference admin, conference add mode, and conference drop mode in the output display.
15.2(4)M	Cisco Unified CME 9.1 Cisco Unified SIP SRST 9.1	This command was modified to include Key Expansion Module (KEM) data in the output display.

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.6.1	Unified SRST 12.0	This command was modified to include the IPv6 address in the output display for Unified SRST.

Examples

Cisco Unified CME

The following is a sample output of the **show voice register pool** command, displaying information for voice register pool 33 in Cisco Unified CME:

```
Router# show voice register pool 33

Pool Tag 33
Config:
 Mac address is 0009.B7F7.532E
 Type is 7960
 Number list 1 : DN 1
 Number list 2 : DN 2
 Number list 3 : DN 3
 Proxy Ip address is 0.0.0.0
 DTMF Relay is disabled
 Call Waiting is enabled
 DnD is disabled
 Busy trigger per button value is 0
 keep-conference is enabled
 template is 1
 Emergency response location 3
 Lpcor Type is local
 Lpcor Incoming is sip_group
 Lpcor Outgoing is sip_group

 Transport type is udp
 service-control mechanism is not supported
 Privacy feature is not configured.
 Privacy button is disabled

Dialpeers created:

Statistics:
 Active registrations : 0

 Total SIP phones registered: 0
 Total Registration Statistics
  Registration requests : 0
  Registration success : 0
  Registration failed : 0
  unRegister requests : 0
  unRegister success : 0
  unRegister failed : 0
```

The following is a sample output of the **show voice register pool** command. The output shows that a meet-me hardware conference administrator has been assigned, the conference creator or any of the participants can add a new participant, and the conference creator can terminate the active video hardware conference by hanging up.

```
Router# show voice register pool 15
 Pool Tag 15
```

```
Config:
  Mac address is 1C17.D340.81F0
  Type is 9951
  Number list 1 : DN 15
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-CP9951/9.0.1
  DTMF Relay is enabled, sip-notify
  Call Waiting is enabled
  DnD is disabled
  Video is enabled
  Camera is enabled
  Busy trigger per button value is 0
  feature-button 5 DnD
  feature-button 6 MeetMe
  keep-conference is enabled
  registration expires timer max is 86400 and min is 60
  template is 1
  kpml signal is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is supported
  registration Call ID is 1c17d340-81f00002-6c48fe8e-03013c10@1.5.40.105
  Registration method: per line
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 3915
  contact IP address: 1.5.40.105 port 5060
  Phone SIS Version:  5.0.0
  GW SIS Version:  1.0.0
  conference admin: yes
  conference add mode: all
  conference drop mode: creator
  paging-dn: config 0 [multicast]  effective 0 [multicast]
...
```

The following is an example of a partial output of the **show voice register pool all** command, showing KEM data with the phone type information:

```
Router# show voice register pool all
Pool Tag 5
Config:
Mac address is B4A4.E328.4698
Type is 9971 addon 1 CKEM
Number list 1 : DN 2
Number list 2 : DN 3
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
Video is enabled
Camera is enabled
Busy trigger per button value is 0
keep-conference is enabled
registration expires timer max is 200 and min is 60
kpml signal is enabled
Lpcor Type is none
```

The following is a sample output of the **show voice register pool all** command, showing the three KEMs configured with phone type 9971:

```
Router# show voice register pool all
Pool Tag 4
Config:
```

```
Mac address is B4A4.E328.4698
Type is 9971 addon 1 CKEM 2 CKEM 3 CKEM
Number list 1 : DN 4
Number list 2 : DN 5
Number list 3 : DN 9
```

Cisco Unified SIP SRST

The following is a sample output of the **show voice register pool** command, displaying all information for voice register pool 1 in Cisco Unified SIP SRST:

```
Router# show voice register pool 1

Pool Tag 1
Config:
 Network address is 192.168.0.0, Mask is 255.255.0.0
 Number list 1 : Pattern is 50.., Preference is 2
 Proxy Ip address is 0.0.0.0
 Default preference is 2
 Incoming called number is
 Translate outgoing called tag is 1
 Class of Restriction List Tag: default
 Incoming corlist name is allowall
 Application is default.new

Dialpeers created:

dial-peer voice 40007 voip
 application default.new
 corlist incoming allowall
 preference 2
 incoming called-number 5001
 destination-pattern 5001
 redirect ip2ip
 session target ipv4:192.168.0.3
 session protocol sipv2
 translate-outgoing called 1
 voice-class codec 1

Statistics:
 Active registrations : 2

 Total Registration Statistics
  Registration requests : 48
  Registration success : 48
  Registration failed : 0
  unRegister requests : 46
  unRegister success : 46
  unRegister failed : 0

Emergency response location 6
```

The following is a sample output of the **show voice register pool brief** command, showing an IPv6 source address configured on a Cisco SIP IP Phone:

```
Router# show voice register pool brief
Pool ID                        IP Address                 Ln DN  Number     State
==== ========================  ========================== == === ========== ============
```

```
1    8.0.0.0                                                           UNREGISTERED
2    2001:420:54FF:13::312:0   2001:420:54FF:13::312:1          10001$     REGISTERED
```

voice class stun usage

The following is a sample output of the **show voice register pool** command, displaying voice-class stun-usage information for voice register pool 51:

```
Router# show voice register pool 51
 Pool Tag 51
Config:
  Mac address is 0011.209F.5D60
  Type is 7960
  Number list 1 : DN 51
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-SIPGateway/IOS-12.x
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  keep-conference is enabled
  template is 10
  Lpcor Type is none

Transport type is udp
  service-control mechanism is not supported
  registration Call ID is 2BA38EE3-17D311DB-800BCD81-A9AD11F0
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 16263646
  contact IP address: 192.168.0.87 port 5060
  Reason for unregistered state:
         No registration request since last reboot/unregister
  voice-class stun-usage is enabled. tag is 1
Dialpeers created:
Dial-peers for Pool 51:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 2
    Registration success   : 2
    Registration failed    : 0
    unRegister requests    : 2
    unRegister success     : 2
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   : 13:43:27.839 IST Tue Apr 20 2010
```

The following table contains descriptions of significant fields shown in the Cisco Unified CME and Cisco Unified SIP SRST output, listed in alphabetical order.

Table 62: show voice register pool Field Descriptions

Field	Description
Active registrations	Shows the current active registrations.
Application	Shows the **application** command configuration for this pool.

Field	Description
Call Waiting	Shows the **call-waiting** command configuration.
Class of Restriction List Tag	Shows the COR tag.
Conference add mode	Shows the current setting of the hardware conference privilege for adding participants.
Conference admin	Shows whether the Cisco Unified SIP IP phone is assigned as the hardware conference administrator or not.
Conference drop mode	Shows who can terminate an active ad-hoc hardware conference by hanging up.
Config	Shows the voice register pool.
Default preference	Shows the default preference value of this pool.
Dialpeers created	Lists all the dial peers created and their contents. Dial-peer contents differ for each application and are not described here.
DnD	Shows the setting of the **dnd-control** command.
DTMF Relay	Shows the setting of the **dtmf-relay** command.
Emergency response location	Shows the ephone's emergency response location to which an emergency response team is dispatched when an emergency call is made.
Incoming called number	Shows the **incoming called-number** command configuration.
Incoming corlist name	Shows the **cor** command configuration.
keep-conference	Shows the status of the **keep-conference** command.
Lpcor Incoming	Shows the setting of the **lpcor incoming** command.
Lpcor Outgoing	Shows the setting of the **lpcor outgoing** command.
Lpcor Type	Shows the setting of the **lpcor type** command.
Mac address	Shows the MAC address of the Cisco Unified SIP IP phone as defined by the **id** command.
Network address and Mask	Shows network address and mask information when the **id** command is configured.
Number list, Pattern, and Preference	Shows the **number** command configuration.
Pool Tag	Shows the assigned tag number of the current pool.
Proxy IP address	Shows the **proxy** command configuration; that is, the IP address of the external SIP server.
Registration failed	Shows the failed registrations.

Field	Description
Registration requests	Shows the incoming registration requests.
Registration success	Shows the successful registrations.
Statistics	Shows the registration statistics for this pool.
Template	Shows the template-tag number for the template applied to the Cisco Unified SIP IP phone.
Total Registration Statistics	Shows the total registration statistics for this pool.
Translate outgoing called tag	Shows the **translate-outgoing** command configuration.
Type	Shows the phone type identified for the Cisco Unified SIP IP phone using the **type** command.
unRegister failed	Reports the number of failed unregisters.
unRegister requests	Shows the incoming unregister/registration expiry requests.
unRegister success	Reports the number of successful unregisters.
Username Password	Shows the values within the authentication credential.

Related Commands

Command	Description
application (voice register pool)	Selects the session-level application for the dial peer associated with an individual Cisco Unified SIP IP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.
call-waiting (voice register pool)	Enables the call-waiting option on a SIP phone.
cor (voice register pool)	Configures a class of restriction on the VoIP dial peers associated with directory numbers.
dnd-control (voice register template)	Enables the Do-Not-Disturb (DND) soft key on SIP phones.
dtmf-relay (voice register pool)	Specifies the list of dual-tone multifrequency (DTMF) relay methods that can be used to relay DTMF audio tones between SIP endpoints.
id (voice register pool)	Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.
incoming called-number (dial peer)	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.
keep-conference (voice register pool)	Allows IP phone conference initiators to exit from conference calls and keep the remaining parties connected.

Command	Description
lpcor incoming	Associates an incoming call with a logical partitioning class of restriction (LPCOR) resource-group policy.
lpcor outgoing	Associates an outgoing call with an LPCOR resource-group policy.
lpcor type	Specifies the LPCOR type for an IP phone.
number (voice register pool)	Indicates the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone.
proxy (voice register pool)	Autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco Unified SIP IP phone registers with a Cisco Unified SIP SRST gateway.
show sip-ua status registrar	Displays all the Cisco Unified SIP IP phones registered with the contact address.
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register dial-peer	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified CME or Cisco Unified SIP SRST register event.
translate-outgoing (voice register pool)	Allows an explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by any Cisco Unified IP phone user.
type (voice register pool)	Defines a phone type for a SIP phone.
voice register pool	Enters voice register pool configuration mode for Cisco Unified SIP IP phones.

show voice register pool after-hour-exempt

To display the details of a phone that has after-hour-exempt enabled on it, use the **show voice register after-hour-exempt** command in privileged EXEC mode.

show voice register after-hour-exempt

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of a phone that has after-hour-exempt enabled. Individual phones can be exempted from call blocking using the after-hour exempt.

Cisco Unified CME

The following is a sample output from this command displaying information for phones with after after-hour-exempt:

```
Router# show voice register pool after-hour-exempt
Pool ID             IP Address      Ln DN  Number                State
==== =============== =============== == === ==================== ============
1    001B.535C.D410  8.3.3.111       3  8                        UNREGISTERED
                                     4  7   451110               UNREGISTERED
2    0015.C68E.6D13                  1  2   45112                UNREGISTERED
3    0021.5553.8998                  1  3   45113                UNREGISTERED
                                     2  3   45113                UNREGISTERED
7    0018.BAC8.D2B1                  1  2   45112                UNREGISTERED
```

Cisco Unified SRST

The following is a sample output from this command displaying information for phones with after after-hour-exempt:

```
Router# show voice register pool after-hour-exempt
Pool ID             IP Address      Ln DN  Number                State
==== =============== =============== == === ==================== ============
1    9.13.18.40      9.13.18.40      1  1   1000                 REGISTERED
                                     2  2   2000                 REGISTERED
                                     3  3   3000                 REGISTERED
                                     4  4   4000                 REGISTERED
                                     5  5   5000                 UNREGISTERED
                                     6  6   6000                 UNREGISTERED
                                     7  7   7000                 UNREGISTERED
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 63: show voice register pool after-hour exempt field descriptions

Field	Description
DN	Directory number of the phone.
IP Address/port	IP address and port number of the phones.
LN	Line number of the phone.
Number	Number of the phones that have after-hour exempt enabled.
Pool	Shows the current pool.
State	Registration state.

Related Commands

Command	Description
after-hour exempt(voice register pool)	Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voice register pool attempted-registrations

To display the details of phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail, use the **show voice register pool attempted-registrations** command in privileged EXEC mode.

show voice register pool attempted-registrations

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of the phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail. If the phone registers successfully after some time, the attempted registration entry will still show up in the attempted-registration table. Use the clear voice register attempted-registrations command to remove the entry from the attempted registration table.

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for show voice register pool attempted-registrations:

```
Router# show voice register pool attempted-registrations
Phones that have attempted registrations and have failed:
 MAC address: 001b.535c.d410
 IP address : 8.3.3.111
 Attempts   : 5
 Time of first attempt : *10:49:51.542 UTC Wed Oct 14 2009
 Time of latest attempt: *10:50:00.886 UTC Wed Oct 14 2009
 Reason for failure    :
         No pool match for the registration request
 MAC address: 0015.c68e.6d13
 IP address : 8.33.33.112
 Attempts   : 4
 Time of first attempt : *10:49:53.418 UTC Wed Oct 14 2009
 Time of latest attempt: *10:50:00.434 UTC Wed Oct 14 2009
 Reason for failure    :
         No pool match for the registration request
 MAC address: 0009.43E9.0B35
 IP address : 9.13.40.83
 Attempts   : 1
 Time of first attempt : *10:49:57.866 UTC Wed Oct 14 2009
 Time of latest attempt: *10:49:57.866 UTC Wed Oct 14 2009
 Reason for failure    :
         No pool match for the registration request
```

The following is a sample output from this command displaying information for show voice register pool attempted-registrations when none of the phones fail:

```
Router# show voice register pool attempted-registrations
 Phones that have attempted registrations and have failed: NONE
```

Related Commands

Command	Description
attempted-registrations size	Allows to set the size of the table that stores information related to SIP phones that attempt to register and fail.
clear voice register attempted-registrations	Clears entries from the attempted-registration table.

show voice register pool cfa

To display the voice register pool details of a phone that has Call Forward All (CFA) enabled, use the **show voice register pool cfa** command in privileged EXEC mode.

show voice register pool cfa

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use this command to display the voice register pool details of the phone with CFA feature enabled. When Call Forward All feature is enabled on Cisco Unified SIP IP phones such as 7940, 7941, 7941GE, 7942, 7945, 7960, 7961, 7961GE, 7962, 7965, 7970, 7971, 7975 through the CFA phone button. The **show voice register pool cfa** command displays only the call forward all B2BUA details.

The **show voice register pool cfa** command also displays the line number and DN number if available under the pool configuration. If call-forward-all is configured under both pool and DN. the configuration under DN takes precedence.

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information:

```
Router# show voice register pool cfa
Pool  Ln  DN  Number  Call Forward All Number
====  ==  ==  ======  ========================
1     2   8           678
      0   1  45111    4555
      4   7  451110   4555
3     1   3  45113    87687
      2   3  45113    87687
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Field	Description
Call Forward All Number	Number to which the calls are forwarded.
DN	Voice register DN tag of the line.
LN	Line number of the telephone number.
Pool	Tag ID of the pool.

Related Commands

Command	Description
call forward b2bua all	Enables call forward all.

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool detail all	Displays the details of all the pools defined in the system.

show voice register pool connected

To display the details of SIP phones that are in connected state, use the **show voice register pool connected** command in privileged EXEC mode.

show voice register pool connected [**brief**]

Syntax Description

brief	(Optional) Displays brief details of SIP phones that are in connected state.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of the phone that are currently in connected state (in conversation). The output for show voice register pool connected command shows details of both calls originating from the SIP phones and calls made towards SIP phones. When used with brief keyword, the show voice register pool connected command displays a brief detail of phones in connected state.

Cisco Unified CME and Cisco Unified SRST

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool connected
Outbound calls from SIP line phones:
Pool tag: 1
==============
 MAC Address    : 001B.535C.D410
 Contact IP     : 8.3.3.111
 Phone Number   : 45111
 Remote Number  : 45112
Call 2
SIP Call ID                : 001b535c-d4100010-79612b5a-336b0db5@8.3.3.111
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 45111
   Called Number           : 45112
   Bit Flags               : 0xC0401C 0x100 0x4
   CC Call ID              : 7
   Source IP Address (Sig ): 8.3.3.5
   Destn SIP Req Addr:Port : [8.3.3.111]:5060
   Destn SIP Resp Addr:Port: [8.3.3.111]:50076
   Destination Name        : 8.3.3.111
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 7
     Stream Type              : voice-only (0)
     Stream Media Addr Type   : 1
     Negotiated Codec         : g729r8 (20 bytes)
     Codec Payload Type       : 18
```

```
      Negotiated Dtmf-relay    : inband-voice
      Dtmf-relay Payload Type  : 0
      QoS ID                   : -1
      Local QoS Strength       : BestEffort
      Negotiated QoS Strength  : BestEffort
      Negotiated QoS Direction : None
      Local QoS Status         : None
      Media Source IP Addr:Port: [8.3.3.5]:17580
      Media Dest IP Addr:Port  : [8.3.3.111]:26298
Options-Ping    ENABLED:NO    ACTIVE:NO
Inbound calls to SIP line phones:

Pool tag: 2
==============
 MAC Address    : 0015.C68E.6D13
 Contact IP     : 8.33.33.112
 Phone Number   : 45112
 Remote Number  : 45111
Call 3
SIP Call ID                : 4DA52F97-ADA311DE-8019803A-FF3E4CBC@8.3.3.5
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 45111
   Called Number           : 45112
   Bit Flags               : 0xC04018 0x100 0x80
   CC Call ID              : 8
   Source IP Address (Sig ): 8.3.3.5
   Destn SIP Req Addr:Port : [8.33.33.112]:5060
   Destn SIP Resp Addr:Port: [8.33.33.112]:5060
   Destination Name        : 8.33.33.112
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 8
     Stream Type              : voice-only (0)
     Stream Media Addr Type   : 1
     Negotiated Codec         : g729r8 (20 bytes)
     Codec Payload Type       : 18
     Negotiated Dtmf-relay    : inband-voice
     Dtmf-relay Payload Type  : 0
     QoS ID                   : -1
     Local QoS Strength       : BestEffort
     Negotiated QoS Strength  : BestEffort
     Negotiated QoS Direction : None
     Local QoS Status         : None
     Media Source IP Addr:Port: [8.3.3.5]:16384
     Media Dest IP Addr:Port  : [8.33.33.112]:30040
```

The following is sample output from this command displaying brief statistical information:

```
Router# show voice register pool connected brief
Pool IP Address      Number               Remote Number
==== =============== ==================== ====================
1    8.3.3.111       45111                45112
Inbound calls to SIP line phones:
Pool IP Address      Number               Remote Number
==== =============== ==================== ====================
2    8.33.33.112     45112                45111
```

Related Commands

Command	Description
show sip-ua calls	Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool ip

To display the details of a SIP phone with a specific IP address, use the **show voice register pool ip** command in privileged EXEC mode.

show voice register pool ip *ip-address*

Syntax Description

ip-address	IPv4 address of the SIP phone .

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of a phone with a specific IP-address. When the pool ID is configured as a mac address or an IP address the registered pools contain the IP address information. The pool information is displayed if the IP addresses match.

When the pool ID is IP and the pool is unregistered, IP address configured under pool is compared with the input IP. When the pool ID is network contact, the IP address of each phone that is registered is compared with the input IP address.

Cisco Unified CME and Cisco Unified SRST

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool ip 8.3.3.111
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
1    001B.535C.D410  8.3.3.111       1  1   45111                REGISTERED
                                     4  7   451110               UNREGISTERED
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 64: show voice register pool ip field descriptions

Field	Description
DN	Voice register DN tag of the line.
ID	Phone identification (ID) address.
IP Address	IP address of the SIP phone.
LN	Line number of the telephone number.
Number	Number of the phones that have a mac address.
Pool	Tag ID of the pool.

Field	Description
State	Registration state of the line.

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool mac

To display the details of voice register pool associated with a specific phone type, use the **show voice register pool mac** command in privileged EXEC mode.

show voice register pool mac H.H.H

Syntax Description

H.H.H	MAC address of the SIP phone attempting to register.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of the phone with the mac address H.H.H. The command displays only the pools that are configured with an ID as mac.

Cisco Unified CME and Cisco Unified SRST

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool mac 001B.535C.D410
Pool ID              IP Address      Ln DN  Number               State
==== ============== =============== == === ==================== ============
1    001B.535C.D410  8.3.3.111       1  1   45111                REGISTERED
                                     4  7   451110               UNREGISTERED
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 65: show voice register pool mac field descriptions

Field	Description
DN	Voice register DN tag of the line.
ID	Phone identification (ID) address.
IP Address	IP address of the SIP phone.
LN	Line number of the telephone number.
Number	Number of the phones that have a mac address.
Pool	Tag ID of the pool.
State	Registration state of the line.

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool on-hold

To display the details of phones that are currently on-hold, use the **show voice register pool oh-hold** command in privileged EXEC mode.

show voice register pool on-hold [**brief**]

Syntax Description

brief	(Optional) Displays brief details of SIP phones that are currently on-hold.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of the phone that are currently on-hold. The show voice register pool on-hold command output also displays a field to show if the hold was a locally initiated hold (initiated on the phone) or if the hold was initiated on the remote end. When used with brief keyword, the show voice register pool on-hold command displays a brief information of the phones that are currently put on hold by the remote caller or have put the remote caller on hold. The “Hold-Origin” field specifies the type of the hold, which can be either remote or local. Local indicates that the call is placed on hold by the local phone and remote indicates that call is placed on hold by the remote phone. In case of double-hold, the hold origin will display the value “Local and Remote”.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for phones ringing in a voice register pool:

```
Router# show voice register pool on-hold brief
Outbound calls from SIP line phones:
Pool IP Address      Number               Remote Number        Hold Origin
==== =============== ==================== ==================== ==============
1    8.3.3.111       45111                45112                Remote & Local
Inbound calls to SIP line phones:
Pool IP Address      Number               Remote Number        Hold Origin
==== =============== ==================== ==================== ==============
2    8.33.33.112     45112                45111                Remote & Local
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for phones on-hold:

```
Router# show voice register pool on-hold
Outbound calls from SIP line phones:
Pool tag: 1
```

```
==============
 MAC Address    : 001B.535C.D410
 Contact IP     : 8.3.3.111
 Phone Number   : 45111
 Remote Number  : 45112
 Local Hold     : CALL HOLD Pressed on SIP Phone
Call 4
SIP Call ID                : 001b535c-d4100010-79612b5a-336b0db5@8.3.3.111
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 45111
   Called Number           : 45112
   Bit Flags               : 0xC0401C 0x10100 0x4
   CC Call ID              : 7
   Source IP Address (Sig ): 8.3.3.5
   Destn SIP Req Addr:Port : [8.3.3.111]:5060
   Destn SIP Resp Addr:Port: [8.3.3.111]:50076
   Destination Name        : 8.3.3.111
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 7
     Stream Type              : voice-only (0)
     Stream Media Addr Type   : 1
     Negotiated Codec         : g729r8 (20 bytes)
     Codec Payload Type       : 18
     Negotiated Dtmf-relay    : inband-voice
     Dtmf-relay Payload Type  : 0
     QoS ID                   : -1
     Local QoS Strength       : BestEffort
     Negotiated QoS Strength  : BestEffort
     Negotiated QoS Direction : None
     Local QoS Status         : None
     Media Source IP Addr:Port: [8.3.3.5]:17580
     Media Dest IP Addr:Port  : [8.3.3.111]:26298
Options-Ping    ENABLED:NO    ACTIVE:NO
Inbound calls to SIP line phones:
Pool tag: 2
==============
 MAC Address    : 0015.C68E.6D13
 Contact IP     : 8.33.33.112
 Phone Number   : 45112
 Remote Number  : 45111
 Remote Hold    : SIP Phone has received CALL HOLD
Call 5
SIP Call ID                : 4DA52F97-ADA311DE-8019803A-FF3E4CBC@8.3.3.5
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 45111
   Called Number           : 45112
   Bit Flags               : 0xC04018 0x4100 0x80
   CC Call ID              : 8
   Source IP Address (Sig ): 8.3.3.5
   Destn SIP Req Addr:Port : [8.33.33.112]:5060
   Destn SIP Resp Addr:Port: [8.33.33.112]:5060
   Destination Name        : 8.33.33.112
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
```

```
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 8
     Stream Type              : voice-only (0)
     Stream Media Addr Type   : 1
     Negotiated Codec         : g729r8 (20 bytes)
     Codec Payload Type       : 18
     Negotiated Dtmf-relay    : inband-voice
     Dtmf-relay Payload Type  : 0
     QoS ID                   : -1
     Local QoS Strength       : BestEffort
     Negotiated QoS Strength  : BestEffort
     Negotiated QoS Direction : None
     Local QoS Status         : None
     Media Source IP Addr:Port: [8.3.3.5]:16384
     Media Dest IP Addr:Port  : [8.33.33.112]:30040
Options-Ping    ENABLED:NO    ACTIVE:NO
```

Related Commands

Command	Description
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show sip-ua calls	Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool phone-load

To display the details of phone-loads associated with phones that are registered to Cisco Unified CME, use the **show voice register pool phone-load** command in privileged EXEC mode.

show voice register pool phone-load

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of the phone-loads associated with phones that are registered with Cisco Unified CME. The phone-load information is taken from the REGISTER message sent by the phone.

Example

The following is a sample output from this command displaying information for voice register pool phone-load:

```
Router# show voice register pool phone-load
Pool  Device Name     Current Version Previous Version
====  ===========     ================ ===================
1     SEP001B535CD410 Cisco-CP7960G/8.0
```

Related Commands

Command	Description
load(voice register global)	Associates a type of Cisco Unified IP phone with a phone firmware file.
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voice register pool registered

To display the details of phones that successfully register to Cisco Unified Communications Manager Express (Cisco Unified CME), use the **show voice register pool registered** command in privileged EXEC mode.

show voice register pool registered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Version	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.
15.2(4)M	Cisco Unified CME 9.1 Cisco Unified SIP SRST 9.1	This command was modified to display Key Expansion Module (KEM) details with the phone type information.

Usage Guidelines

Use the **show voice register pool registered** command to display the details of phones that are successfully registered to Cisco Unified CME and Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST).

Cisco Unified CME

The following is a sample output displaying information for a registered voice register pool in Cisco Unified CME:

```
Router# show voice register pool registered
Pool Tag 1
Config:
  Mac address is 001B.535C.D410
  Type is 7960
  Number list 1 : DN 1
  Number list 3 : DN 8
  Number list 4 : DN 7
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-CP7960G/8.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  call-forward phone all is 4566
  call-forward b2bua all 4555
  keep-conference is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is supported
  registration Call ID is 001b535c-d410790d-17a6877e-5d04bbc5@8.3.3.111
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 45111
  contact IP address: 8.3.3.111 port 5060
Dialpeers created:
```

```
Dial-peers for Pool 1:
dial-peer voice 40001 voip
 destination-pattern 45111
 session target ipv4:8.3.3.111:5060
 session protocol sipv2
  call-fwd-all         4555
  after-hours-exempt   FALSE
Statistics:
  Active registrations  : 1
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 1
    Registration success   : 1
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   : *11:40:32.263 UTC Wed Oct 14 2009
    Last unregister request time :
    Register success time        : *11:40:32.267 UTC Wed Oct 14 2009
    Unregister success time      :
```

The following is a sample output displaying information for a registered voice register pool with a Cisco Unified 9971 Session Initiation Protocol (SIP) IP phone attached to a Cisco SIP IP Phone CKEM 36-Button Line Expansion Module:

```
Router# show voice register pool registered
Pool Tag 5
Config:
Mac address is B4A4.E328.4698
Type is 9971 addon 1 CKEM
Number list 1 : DN 2
Number list 2 : DN 3
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
Video is enabled
Camera is enabled
Busy trigger per button value is 0
keep-conference is enabled
registration expires timer max is 200 and min is 60
kpml signal is enabled
Lpcor Type is none
```

Cisco Unified SRST

```
The following is a sample output displaying information for a registered voice register
pool in Cisco Unified SRST:
Router# show voice register pool registered
 Pool Tag 1
Config:
  Ip address is 9.13.18.40, Mask is 255.255.0.0
  Number list 1 : DN 1
  Number list 2 : DN 2
  Number list 3 : DN 3
  Number list 4 : DN 4
  Number list 5 : DN 5
```

```
  Number list 6 : DN 6
  Number list 7 : DN 7
  Proxy Ip address is 0.0.0.0
  DTMF Relay is enabled, rtp-nte, sip-notify
  kpml signal is enabled
  Lpcor Type is none
Dialpeers created:
Dial-peers for Pool 1:
dial-peer voice 40004 voip
 destination-pattern 1000
 redirect ip2ip
 session target ipv4:9.13.18.40:19633
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
dial-peer voice 40001 voip
 destination-pattern 2000
 redirect ip2ip
 session target ipv4:9.13.18.40:19634
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
dial-peer voice 40002 voip
 destination-pattern 3000
 redirect ip2ip
 session target ipv4:9.13.18.40:19635
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
dial-peer voice 40003 voip
 destination-pattern 4000
 redirect ip2ip
 session target ipv4:9.13.18.40:19636
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
Statistics:
  Active registrations  : 4
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 4
    Registration success   : 4
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   : .05:22:55.604 UTC Tue Oct 6 2009
    Last unregister request time :
    Register success time        : .05:22:55.604 UTC Tue Oct 6 2009
    Unregister success time      :
```

The following table contains descriptions of significant fields shown in the **show voice register pool registered** command output, listed in alphabetical order.

Table 66: show voice register pool registered Field Descriptions

Field	Description
Active registrations	Shows the current active registrations.
Application	Shows the **application** command configuration for this pool.
Call Waiting	Shows the setting of the **call-waiting** command.
Class of Restriction List Tag	Shows the COR tag.
Config	Shows the voice register pool.
Current phone-load	Shows the current version of the phone load.
Default preference	Shows the default preference value of this pool.
Dialpeers created	Results in a list of all dial peers created and their contents. Dial-peer contents differ for each application and are not described here.
DnD	Shows the setting of the **dnd-control** command.
DTMF Relay	Shows the setting of the **dtmf-relay** command.
Emergency response location	Shows the ephone’s emergency response location to which an emergency response team is dispatched when an emergency call is made.
Incoming called number	Shows the **incoming called-number** command configuration.
Incoming corlist name	Shows the **cor** command configuration.
keep-conference	Shows the status of the **keep-conference** command.
Lpcor Incoming	Shows the setting of the **lpcor incoming** command.
Lpcor Outgoing	Shows the setting of the **lpcor outgoing** command.
Lpcor Type	Shows the setting of the **lpcor type** command.
Mac address	Shows the MAC address of this SIP phone as defined by the **id** command.
Network address and Mask	Shows network address and mask information when the **id** command is configured.
Number list, Pattern, and Preference	Shows the **number** command configuration.
Pool Tag	Shows the assigned tag number of the current pool.
Previous phone-load	Shows the version of the previous phone load.
Proxy IP address	Shows the **proxy** command configuration; that is, the IP address of the external SIP server.

Field	Description
Registration failed	Shows the failed registrations.
Registration requests	Shows the incoming registration requests.
Registration success	Shows the successful registrations.
Statistics	Shows the registration statistics for this pool.
statistics time-stamps	Shows the registration statistics for this pool with specific time stamps.
Template	Shows the template-tag number for the template applied to this SIP phone.
Total Registration Statistics	Shows the total registration statistics for this pool.
Translate outgoing called tag	Shows the **translate-outgoing** command configuration.
Type	Shows the phone type identified for this SIP phone using the **type** command.
unRegister failed	Reports the number of failed unregisters.
unRegister requests	Shows the incoming unregister/registration expiry requests.
unRegister success	Reports the number of successful unregisters.
Username Password	Shows the values within the authentication credential.

Related Commands

Command	Description
application (voice register pool)	Selects the session-level application for the dial peer associated with an individual Cisco Unified SIP IP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.
call-waiting (voice register pool)	Enables the call-waiting option on a SIP phone.
cor (voice register pool)	Configures a class of restriction on the VoIP dial peers associated with directory numbers.
dnd-control (voice register template)	Enables the Do-Not-Disturb (DND) soft key on SIP phones.
dtmf-relay (voice register pool)	Specifies the list of dual-tone multifrequency (DTMF) relay methods that can be used to relay DTMF audio tones between SIP endpoints.
id (voice register pool)	Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.
incoming called-number (dial peer)	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.

Command	Description
keep-conference (voice register pool)	Allows IP phone conference initiators to exit from conference calls and keep the remaining parties connected.
lpcor incoming	Associates an incoming call with a logical partitioning class of restriction (LPCOR) resource-group policy.
lpcor outgoing	Associates an outgoing call with an LPCOR resource-group policy.
lpcor type	Specifies the LPCOR type for an IP phone.
number (voice register pool)	Indicates the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone.
proxy (voice register pool)	Autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco Unified SIP IP phone registers with a Cisco Unified SIP SRST gateway.
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register dial-peers	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified CME register event.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool unregistered	Displays the details of voice register pools that do not have any phones registered.
translate-outgoing (voice register pool)	Allows an explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by any Cisco Unified IP phone user.
type (voice register pool)	Defines a phone type for a SIP phone.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voice register pool remote

To display the details of phones that are at a remote location, use the **show voice register pool remote** command in privileged EXEC mode.

show voice register pool remote

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of the phones that are at remote location and do not have an address resolution protocol (ARP) entry. If the pool id is MAC or IP, the entire pool detail is displayed in a brief format. If the pool id is network, only the line details with remote contact IP address are displayed. In Cisco Unified SRST, if the pool id is IP and if the pool is not registered, the configured IP is checked to see if it is a remote IP.

Cisco Unified CME

The following is a sample output from this command displaying information for remote phones:

```
Router# show voice register pool remote
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
1    001B.535C.D410 8.3.3.111        1  1  45111                 REGISTERED
                                     3  8                        UNREGISTERED
                                     4  7  451110                UNREGISTERED
2    8.3.3.112                       1  2  45112                 REGISTERED
3    8.3.0.0                         1  3  45113                 REGISTERED
```

Cisco Unified SRST

The following is a sample output from this command displaying information for remote phones:

```
Router# show voice register pool remote
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
1    001B.535C.D410  8.33.33.111      1   1  45111               REGISTERED
                                      3   8                      UNREGISTERED
                                      4   7  451110              UNREGISTERED
2    8.33.33.112     8.33.33.112      1   2  45112               REGISTERED
3    8.3.0.0         8.3.44.116       1   3  45113               REGISTERED
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Related Commands

Command	Description
show voice register all voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register dial-peer	Displays details of all dynamically created VoIP dial peers associated with the Cisco SIP SRST or Cisco CME register event.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voice register pool ringing

To display the details of phones that are currently in ringing state, use the **show voice register pool ringing** command in privileged EXEC mode.

show voice register pool ringing [**brief**]

Syntax Description

brief	(Optional) Displays brief details of SIP phones that are currently in ringing state.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of the phone that are currently in ringing state. When used with the brief keyword, the show voice register pool ringing brief command only displays information related to calls that are bound towards the SIP phones.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for phones ringing in a voice register pool:

```
Router# show voice register pool ringing brief
Pool IP Address      Number               Remote Number
==== =============== ==================== ====================
2    8.33.33.112     45112                45111
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for phones ringing in a voice register pool:

```
Router# show voice register pool ringing
Pool tag: 2
==============
 MAC Address    : 0015.C68E.6D13
 Contact IP     : 8.33.33.112
 Phone Number   : 45112
 Remote Number  : 45111
Call 1
SIP Call ID                : C0B5DA7-ADA311DE-8011803A-FF3E4CBC@8.3.3.5
   State of the call       : STATE_RECD_PROCEEDING (4)
   Substate of the call    : SUBSTATE_PROCEEDING_PROCEEDING (2)
   Calling Number          : 45111
   Called Number           : 45112
   Bit Flags               : 0xC00018 0x100 0x280
   CC Call ID              : 5
```

```
Source IP Address (Sig ): 8.3.3.5
Destn SIP Req Addr:Port : [8.33.33.112]:5060
Destn SIP Resp Addr:Port: [8.33.33.112]:5060
Destination Name        : 8.33.33.112
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object         : 0x0
Media Mode              : flow-through
Media Stream 1
  State of the stream      : STREAM_ACTIVE
  Stream Call ID           : 5
  Stream Type              : voice+dtmf (1)
  Stream Media Addr Type   : 1
  Negotiated Codec         : No Codec    (0 bytes)
  Codec Payload Type       : 255 (None)
  Negotiated Dtmf-relay    : inband-voice
  Dtmf-relay Payload Type  : 0
  QoS ID                   : -1
  Local QoS Strength       : BestEffort
  Negotiated QoS Strength  : BestEffort
  Negotiated QoS Direction : None
  Local QoS Status         : None
  Media Source IP Addr:Port: [8.3.3.5]:16882
```

Related Commands

Command	Description
show sip-ua calls	Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool telephone-number

To display the details of a phone line with a specific telephone-number, use the **show voice register pool telephone-number** command in privileged EXEC mode.

show voice register pool telephone-number number

Syntax Description

number	Number identifying a specific phone.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of the phone line with the specified telephone-number. If the line is registered, the contact ip address will be displayed. When the phone line is not registered and the pool ID type is network IP, the IP address is not displayed. When the phone line is not registered but some other line is registered for the same pool with MAC or IP address, then the IP address is displayed.

Cisco Unified CME

The following is a sample output from this command displaying all statistical information:

```
Router# show voice register pool telephone number 45112
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
2    0015.C68E.6D13                  1  2   45112                UNREGISTERED
7    0018.BAC8.D2B1                  1  2   45112                UNREGISTERED
```

Cisco Unified SRST

```
Router# show voice register pool telephone-number 1000
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
1    9.13.18.40      9.13.18.40      1  1   1000                 REGISTERED
```

The table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 67: show voice register pool telephone number field descriptions

Field	Description
DN	Directory number of the phone.
ID	Phone identification (ID) address.
IP Address	IP address and port number of the phones

Field	Description
LN	Line number of the phone.
Number	Number of the phones.
Pool	Shows the current pool.
State	Registration state.

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool detail all	Displays the details of all the pools defined in the system.

show voice register pool type

To display the details of voice register pools associated with a specific phone type, use the **show voice register pool type** command in privileged EXEC mode.

show voice register pool type *type*

Syntax Description

type	3911, 3951, 7905, 7906, 7911, 7912, 7940, 7941, 7941GE, 7942, 7945, 7960, 7961, 7961GE, 7962, 7965, 7970, 7971, 7975, 7800 Series, 8800 Series, ATA (Cisco SIP Phone ATA), ATA-191, CKEM (Cisco SIP Key Expansion Module), CP-8800-Audio (Cisco SIP Key Expansion Module), CP-8800-Video (Cisco SIP Key Expansion Module), P100 (PingTel Xpressa 100), P600 (Polycom SoundPoint 600).

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1	This command was introduced.
15.2(4)M	Cisco Unified CME 9.1	This command was modified to add CKEM as a value for the *type* argument to display the details of voice register pools associated with all the phones configured with KEMs.
15.3(3)M	Cisco Unified CME 10.0	This command was enhanced to display the properties for new sip phone models configured using SIP fast track feature. New keyword option **all** was added to display all the phone models being used in the system along with the associated pools and registration details.
Cisco IOS XE Gibraltar 16.10.1a	Unified CME 12.5	This command was modified to add **CP-8800-Audio** and **CP-8800-Video** as values for the *type* argument to display the details of voice register pools associated with A-KEMs and V-KEMs.
Cisco IOS XE Gibraltar 16.10.1a	Unified CME 12.5	This command was modified to add **ATA-191** as the value for the *type* argument to display the details of voice register pools associated with Cisco ATA 191.

Usage Guidelines

Use the **show voice register pool type** command to display the details of voice register pools associated with a specific phone type.

The **show voice register pool type** command only takes the configured value of the phone type into consideration.

The CKEM value is available for Cisco Unified CME only and is not available for Cisco Unified SRST.

Examples

The following is a sample output of the **show voice register pool type** command for a Cisco Unified 7960 SIP IP phone, displaying all statistical information:

```
Router# show voice register pool type 7960
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
1    001B.535C.D410  8.3.3.111       1  1   45111                REGISTERED
                                     4  7   451110               UNREGISTERED
2    0015.C68E.6D13                  1  2   45112                UNREGISTERED
```

The following is a sample output of the **show voice register pool type** command, showing all the phones configured with CP-8800-Audio:

```
Router# show voice register pool type CP-8800-Audio
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
2    38ED.18AF.8993  8.55.0.199      1   2   7001$               REGISTERED
```

The following is a sample output of the **show voice register pool type** command, showing all the 8865 phones configured with CP-8800-Video:

```
Router# show voice register pool type CP-8800-Video
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
4    00CC.FC99.8973  8.55.0.88      1   1   5001$                REGISTERED
```

The following is a sample output of the **show voice register pool type** command, showing all the phones configured with CKEM:

```
Router# show voice register pool type CKEM
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
4    B4A4.E328.4698  9.45.31.111     1  4   5589$                REGISTERED
```

The following is a sample output of the **show voice register pool type** command for a Cisco Unified 7821 SIP IP phone configured using SIP fast track feature, displaying all statistical information:

```
Router# show voice register pool type 7821
FastTrack Phone Model : 7821
Pooltype(index) representing the phone model : 48
Reference pooltype to inherit the properties from : 6921
Number of lines supported : 2 (inherited from 6921)
Number of addon modules supported : 0 (inherited from 6921)
Default session transport : UDP (inherited from 6921)
Description(helpstring) : Cisco IP Phone 7821
Phone supports GSM : NO (inherited from 6921)
Phone supports Telnet acess : NO (inherited from 6921)
Phone supports firmware download from CME : YES (inherited from 6921)
Phone specific XML tags :
<maxNumCalls>12</maxNumCalls> (inherited from 6921)
<busyTrigger>12</busyTrigger> (inherited from 6921)
Phone family : RTL_PHONES
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
6    D824.BD27.9EAC  9.44.29.44      1  6   4080$                REGISTERED
```

The following is a sample output of the **show voice register pool type** all command, showing all the phone models used in the system:

```
Router# show voice register pool type all
Builtin Phone Model : 9971
```

```
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
3    A418.7529.93B0  9.44.29.41      1  3   4012$                REGISTERED
9    001E.7A25.D4EE                  1  9   4006                 UNREGISTERED

Builtin KEM Module : CKEM
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
8    1234.1234.1234                                              UNREGISTERED

Builtin Phone Model : Jabber-MAC
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
1    0021.5553.19D1                  1  1   4010                 UNREGISTERED

FastTrack Phone Model : 8900
Pooltype(index) representing the phone model : 52
Reference pooltype to inherit the properties from : 8945
Number of lines supported : 4
Number of addon modules supported : 0 (inherited from 8945)
Default session transport : UDP (inherited from 8945)
Description(helpstring) : Cisco SIP Phone 8945
Phone supports GSM : NO (inherited from 8945)
Phone supports Telnet acess : NO (inherited from 8945)
Phone supports firmware download from CME : YES
Phone spcific XML tags :
<maxNumCalls>24</maxNumCalls>
<busyTrigger>24</busyTrigger>
Phone family : GUMBO_PHONES
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
7    D824.BD27.9EBD  9.44.29.45      1  7   4022$                REGISTERED

Router#
```

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool type summary

To display the total count of registered and unregistered phones for each Session Initiation Protocol (SIP) phone type , use the **show voice register pool type summary** command in privileged EXEC mode.

show voice register pool type summary

Syntax Description

This command has no arguments or keywords.

Command Default

This command has no default behavior or values.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3) M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

Use this command to view the count of the phones configured, registered and unregistered in the SIP mode.

Example

The following is a sample output of the **show voice register pool type summary** command:

```
router# show voice register pool type summary
=============================================================
PhoneType              Configured   Registered   Unregistered
=============================================================
7970                        1            1             0
8941                        4            3             1
Unknown Phone type          4            0             4
=============================================================
Total Phones                9            4             5
=============================================================
```

show voice register pool unregistered

To display the details of the voice registration pools that do not have any phones registered, use the **show voice register pool unregistered** command in privileged EXEC mode.

show voice register pool unregistered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to display the details of the pools that do not have any active registrations. In Cisco Unified SRST, if multiple phones are trying to register through the same pool and if one phone successfully registers and the others do not, the pool is not considered as an unregistered pool, as it does have an active registration of the registered phone.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for pools with no active registeration:

```
Router# show voice register pool unregistered
Pool Tag: 2
 MAC Address                : 0015.C68E.6D13
 No. of attempts to register: 0
 Unregister time            :
 Last register request time :
 Reason for state unregister:
         No registration request since last reboot/unregister
 Pool Tag: 3
 MAC Address                : 0021.5553.8998
 No. of attempts to register: 0
 Unregister time            :
 Last register request time :
 Reason for state unregister:
         No registration request since last reboot/unregister
 Pool Tag: 4
 MAC Address                : 8989.9867.8769
 No. of attempts to register: 0
 Unregister time            :
 Last register request time :
 Reason for state unregister:
         No registration request since last reboot/unregister
```

Related Commands

Command	Description
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool registered	Displays details of phones that sucessfully register to Cisco Unified CME or Cisco Unified SRST.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voice register profile

To display the content of configuration files that are in ASCII text format, use the **show voice register profile** command in privileged EXEC mode.

show voice register profile text *tag*

Syntax Description

tag	Unique identifier for voice register profile to be displayed. Range is 1–500.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

Use this command to display ASCII configuration files for the Cisco IP Phone 7905 and 7905G, Cisco IP Phone 7912 and 7912G, Cisco ATA-186, or Cisco ATA-188. To generate ASCII text files, use the **file text** command.

Examples

The following is sample output from this command displaying information in the configuration profile for voice register pool 4:

```
Router# show voice register profile text 4
 Pool Tag: 4
#txt
 AutoLookUp:0
 DirectoriesUrl:0
...
 CallWaiting:1
 CallForwardNumber:0
 Conference:1
 AttendedTransfer:1
 BlindTransfer:1
...
 SIPRegOn:1
 UseTftp:1
 UseLoginID:0
 UIPassword:0
 NTPIP:0.0.0.0
 UID:2468
...
```

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 68: show voice register profile Field Descriptions

Field	Description
Attended Transfer	Setting of soft key for attended transfer in a SIP phone template as defined by using the **transfer-attended** command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.

Field	Description
Auto Lookup	1 indicates that Auto Lookup is enabled. 0 indicates that it is disabled.
Blind Transfer	Setting of soft key for blind transfer in a SIP phone template as defined by using the **transfer-blind** command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.
Call Waiting	Setting of the call-waiting option on a SIP phone as defined by using the **call-waiting** command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.
Call Forward Number	Number to which incoming calls are forwarded
Conference	Setting of soft key for conference in a SIP phone template as defined by using the **conference** command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.
Directories URL	1 indicates that the Directories feature button for the phone is enabled. 0 indicates that it is disabled.
NTPIP	IP address for the NTP source
Pool tag	Pool tag of the configuration file being requested.
SIP Reg On	1 indicates that the registration with external proxy server for the phone is enabled. 0 indicates that it is disabled.
UI Password	1 indicates that the UI password is enabled on the phone. 0 indicates that dit is disabled.
UID	Authenticatuion credential for SIP phone.
Use Login ID	1 indicates that “use login id” for phone is enabled. 0 indicates that it is disabled.

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phone.
file text (voice register global)	Generates ASCII text files for the Cisco IP Phone 7905 and 7905G, Cisco IP Phone 7912 and 79012G, Cisco ATA-186, or Cisco ATA-188.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

show voice register session-server

To display the call details of the registered session servers, use the **show voice register session-server** command in privileged EXEC mode.

show voice register session-server

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
12.4(22)T	This command was introduced in a release earlier than Cisco IOS Release 12.4(22)T.

Examples

The following is sample output from the **show voice register session-server** command:

```
Router# show voice register session-server
Feature server 2, keepalive 60, register-uri CISCO-80NVCGATW_1259887561000
Session reg_number 569, refID 9B2783C0
Route point voice_reg_pool 28 reg_number 570
Route point voice_reg_pool 9 reg_number 571
Route point voice_reg_pool 19 reg_number 572
Route point voice_reg_pool 22 reg_number 573
Subscription sub_id 1133, calledNumber 1242
Subscription sub_id 1135, calledNumber 1054
Subscription sub_id 1138, calledNumber 1155
Subscription sub_id 1140, calledNumber 1188
Subscription sub_id 1142, calledNumber 261
Subscription sub_id 1146, calledNumber 1055
Subscription sub_id 1147, calledNumber 1100
Subscription sub_id 1149, calledNumber 1025
Subscription sub_id 1152, calledNumber 264
Subscription sub_id 1154, calledNumber 267
Subscription sub_id 1156, calledNumber 1185
Subscription sub_id 1157, calledNumber 1218
Subscription sub_id 1160, calledNumber 1056
Subscription sub_id 1161, calledNumber 263
Subscription sub_id 1163, calledNumber 1186
Subscription sub_id 1165, calledNumber 1243
Subscription sub_id 1167, calledNumber 1053
Subscription sub_id 1169, calledNumber 1120
Subscription sub_id 1171, calledNumber 1154
Subscription sub_id 1173, calledNumber 265
```

The following table describes the significant fields shown in this output.

Table 69: show voice register session-server Field Descriptions

Field	Definition
Feature server	The number of active feature servers.
keepalive	Interval, in seconds, at which the peer sends keepalive messages. The range is from 1 to 60 seconds. The default is 60 seconds.

Field	Definition
register-uri	The registered Uniform Resource Identifier (URI) for the server.
Session reg_number	The registered number of the session.
Route point voice_reg_pool	Denotes the registered virtual device for application redirection.
Subscription sub_id	The subidentification number of the subscription.

show voice register statistics

To display statistics associated with the registration event, use the **show voice register statistics** command in privileged EXEC mode.

show voice register statistics [{**global** | **pool** **tag**}]

Syntax Description

global	(Optional) Displays aggregate statistics associated with the SIP phone registration event.
pool tag	(Optional) Displays registration pool statistics associated with a specific pool tag. The maximum number of pools is version and platform dependent. Type **?** to display a list of values.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.
15.1(2)T	Cisco CME 8.1 Cisco SIP SRST 8.1	This command was modified. The global and pool keywords and tag argument were added. The output display was also modified to show more information about pools in unregistered state and time-stamps of registration event.

Usage Guidelines

When using the **show voice register statistics** command, you can verify that the number of Registration and unRegister successes for global statistics are the sum of the values in the individual pools. Because some Registrations fail even before matching a voice register pool, for Registration and unRegister failed statistics the value is not the sum of the values in the individual pools. Immediate failures are accounted in the global statistics.

In Cisco Unified CME 8.1 and Cisco Unified SIP SRST 8.1, the time-stamps for the events is displayed along with other registration related statistics. The command output also displays the reason for pools in unregistered state. Use the show voice register statistics command with pool tag keyword to display registration pool statistics associated with a specific pool.

When using the global keyword, the show voice register command output displays the aggregate statistics associated with SIP phone registration. The output of this command also displays the attempted-registrations table.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information:

```
Router# show voice register statistics
Sample Output:
Global statistics
  Active registrations  : 2
  Total SIP phones registered: 2
  Total Registration Statistics
    Registration requests  : 3
    Registration success   : 2
    Registration failed    : 1
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
    after last unregister        : 1
    Last Register Request Time   : *11:42:31.783 UTC Wed Sep 16 2009
    Last Unregister Request Time :
    Register Success Time        : *11:11:56.707 UTC Wed Sep 16 2009
    Unregister Success Time      :
Register pool 1 statistics
  Active registrations  : 1
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 1
    Registration success   : 1
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
    after last unregister        : 0
    Last Register Request Time   : *11:11:54.615 UTC Wed Sep 16 2009
    Last Unregister Request Time :
    Register Success Time        : *11:11:54.623 UTC Wed Sep 16 2009
    Unregister Success Time      :
Register pool 2 statistics
  Active registrations  : 1
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 1
    Registration success   : 1
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
    after last unregister        : 0
    Last Register Request Time   : *11:11:56.707 UTC Wed Sep 16 2009
    Last Unregister Request Time :
    Register Success Time        : *11:11:56.707 UTC Wed Sep 16 2009
    Unregister Success Time      :
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information:

```
Router# show voice register statistics global
Global Statistics:
  Active registrations  : 1
```

```
  Total SIP phones registered: 2
  Total Registration Statistics
    R egistration requests  : 97715
    Registration success   : 3
    Registration failed    : 97712
    unRegister requests    : 1
    unRegister success     : 1
    unRegister failed      : 0
    Attempts to register
           after last unregister : 97712
    Last register request time   : *06:45:11.127 UTC Wed Oct 14 2009
    Last unregister request time : *11:56:22.179 UTC Tue Oct 13 2009
    Register success time        : *12:10:37.263 UTC Tue Oct 13 2009
    Unregister success time      : *11:56:22.182 UTC Tue Oct 13 2009
 Phones that have attempted registrations and have failed:
 MAC address: 001b.535c.d410
 IP address : 8.3.3.111
 Attempts   : 97712
 Time of first attempt : *12:20:32.775 UTC Tue Oct 13 2009
 Time of latest attempt: *06:46:14.815 UTC Wed Oct 14 2009
 Reason for failure    :
         Unauthorized registration request
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information associated with pool 1:

```
Router# show voice register statistics pool 1
Pool 1 Statistics:
  Active registrations  : 0
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 2
    Registration success   : 2
    Registration failed    : 0
    unRegister requests    : 1
    unRegister success     : 1
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   : *12:10:37.259 UTC Tue Oct 13 2009
    Last unregister request time : *11:56:22.179 UTC Tue Oct 13 2009
    Register success time        : *12:10:37.263 UTC Tue Oct 13 2009
    Unregister success time      : *11:56:22.182 UTC Tue Oct 13 2009
  Reason for unregistered state:
         No registration request since last reboot/unregister
```

The following table describes the significant fields shown in this output.

Table 70: show voice register statistics Field Descriptions

Field	Description
Statistics:	Used with the **all**, **pool**, and **statistics** keywords. Shows the registration statistics for this pool.

Field	Description
Active registrations	Used with the **all**, **pool**, and **statistics** keywords. Shows the current active registrations.
Last Register Request Time	Used with all, pool, and statistics keywords. Shows details such as day, date, and time when the phones requested to register the last time.
Last unRegister Request Time	Used with all, pool, and statistics keywords. Shows details such as day, date, and time when the phones requested to unregister the last time.
Total Registration Statistics	Used with the **all**, **pool**, and **statistics** keywords. Shows the total registration statistics for this pool.
Registration requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming registration requests.
Registration success	Used with the **all**, **pool**, and **statistics** keywords. Shows the successful registrations.
Registration failed	Used with the **all**, **pool**, and **statistics** keywords. Shows the failed registrations.
unRegister requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming unregister/registration expire requests.
unRegister success	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of successful unregisters.
unRegister failed	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of failed unregisters.
Global statistics	Used with the **statistics** keyword. Details all active registrations.
Register pool *number* statistics	Used with the **statistics** keyword. Details specific pool statistics.

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool attempted-registrations	Displays the details of phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail.

show voice register template

To display all configuration information associated with a Cisco Unified SIP IP phone template, use the **show voice register template** command in privileged EXEC mode.

show voice register template {*template-tag* | **all**}

Syntax Description

template-tag	Number of the template for which to display information. Range is 1 to 5.
all	Displays all configuration information associated with all the Cisco Unified SIP IP phone templates.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was modified to include emergency response location (ERL) information assigned to a Cisco Unified SIP IP phone in the output display.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified to include logical partitioning class of restriction (LPCOR) information in the output display.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. All keyword was added. Pools that have the template defined are also displayed in the output. Voice-class stun-usage information is displayed in the output.
15.2(2)T	Cisco Unified CME 9.0	This command was modified to include conference admin, conference add mode, and conference drop mode in the output display.

Usage Guidelines

Use the **show voice register template** command to display all configuration information associated with a Cisco Unified SIP IP phone template defined in a system. Use the **all** keyword with the **show voice register template** command to display the details of all the templates defined in the system. A maximum of 10 templates can be configured and hence, the details of a maximum of 10 templates are displayed in the output.

Examples

The following is a sample output from the **show voice register template** command displaying information for a voice register template:

```
Router# show voice register template 1

Temp Tag 1
Config:
  Attended Transfer is enabled
  Blind Transfer is enabled
  Semi-attended Transfer is enabled
  Conference is enabled
  Caller-ID block is disabled
  DnD control is enabled
  Anonymous call block is disabled
  Voicemail is 56789, timeout 15
  softkey connected  Confrn Endcall Hold Trnsfer
  softkey hold  Newcall Resume
  softkey idle  Cfwdall Newcall Redial
  softkey seized  Cfwdall Endcall Redial
  Emergency response location 6
  Lpcor type local
  Lpcor incoming sccp_phone1
  Lpcor outgoing sccp_phone1
```

The following is a sample output from the **show voice register template** command displaying voice-class stun-usage information for voice register template 10:

```
Router# show voice register template 10
Temp Tag 10
Config:
  Attended Transfer is enabled
  Blind Transfer is enabled
  Semi-attended Transfer is enabled
  Conference is enabled
  Caller-ID block is disabled
  DnD control is enabled
  Anonymous call block is disabled
  softkey connected  Park Confrn Endcall Hold Trnsfer
  voice-class stun-usage is enabled. tag is 1
  Lpcor type none
  Pool 2 has this template configured
  Pool 3 has this template configured
  Pool 5 has this template configured
  Pool 6 has this template configured
  Pool 7 has this template configured
  Pool 8 has this template configured
  Pool 9 has this template configured
  Pool 10 has this template configured
  Pool 11 has this template configured
  Pool 50 has this template configured
```

The following is a sample output from the **show voice register template** command. The output shows that a hardware conference administrator has been assigned, only the conference creator can add a new participant, and the conference creator can terminate the active video hardware conference by hanging up.

```
Router# show voice register template 5
Temp Tag 5
Config:
  Attended Transfer is enabled
  Blind Transfer is enabled
  Semi-attended Transfer is enabled
  Conference softkey is enabled
```

```
Caller-ID block is disabled
DnD control is enabled
Video is disabled
Camera is enabled
Anonymous call block is disabled
Lpcor type none
paging-dn 0 [multicast]
conference admin: yes
conference add mode: creator
```

conference drop mode: creator

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 71: show voice register template Field Descriptions

Field	Description
Anonymous call block	Status of anonymous caller blocking defined with the **anonymous block** command.
Attended Transfer	Status of attended transfer soft key defined with the **transfer-attended** command.
Blind Transfer	Status of blind transfer soft key defined with the **transfer-blind** command.
Caller-ID block	Status of caller-id feature defined with the **caller-id block** command.
Conference	Status of conference soft key defined with the **conference** command.
Conference admin	Shows whether the Cisco Unified SIP IP phone is assigned as the hardware conference administrator or not.
Conference add mode	Current setting of hardware conference privilege for adding participants.
Conference drop mode	Shows who can terminate an active ad-hoc hardware conference by hanging up.
Config:	List of configuration options defined for this template.
Dnd controls	Status of Do-Not-Disturb soft key defined with the **dnd-control** command.
Emergency response location	The ephone's emergency response location to which an emergency response team is dispatched when an emergency call is made.
Lpcor incoming	Setting of the **lpcor incoming** command.
Lpcor outgoing	Setting of the **lpcor outgoing** command.
Lpcor type	Setting of the **lpcor type** command.
Temp Tag	Tag number of the requested template.
VAD	Status of voice activity detection defined with the **vad** command.
Voicemail	Voice-mail extension and timeout value defined with the **voice-mail** command.

Related Commands

Command	Description
show voice register all	Displays all voice register information, including statistics, pools, and dial peers.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.

show voice register tftp-bind

To display the current configuration files accessible to SIP phones, use the **show voice register tftp-bind** command in privileged EXEC mode.

show voice register tftp-bind

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command provides a list of configuration files that are accessible to SIP phones using TFTP.

Examples

The following is sample output from this command:

```
Router(config)# show voice register tftp-bind
 tftp-server SIPDefault.cnf url system:/cme/sipphone/SIPDefault.cnf
 tftp-server syncinfo.xml url system:/cme/sipphone/syncinfo.xml
 tftp-server SIP0009B7F7532E.cnf url system:/cme/sipphone/SIP0009B7F7532E.cnf
 tftp-server SIP000ED7DF7932.cnf url system:/cme/sipphone/SIP000ED7DF7932.cnf
 tftp-server SIP0012D9EDE0AA.cnf url system:/cme/sipphone/SIP0012D9EDE0AA.cnf
 tftp-server gk123456789012 url system:/cme/sipphone/gk123456789012
 tftp-server gk123456789012.txt url system:/cme/sipphone/gk123456789012.txt
```

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 72: show voice register tftp-bind Field Descriptions

Field	Description
ata<mac-address>	Cisco SIP configuration profile for a particular Cisco ATA-186 or Cisco ATA-188 as indicated by the <mac-address>. This file is generated by using the **create profile command.**
ata<mac-address>.txt	ASCII text file of a Cisco SIP configuration profile for a particular Cisco ATA-186 or Cisco ATA-188 as indicated by the <mac-address>. This file is generated by using the **file text command.**
gk<mac-address>	Cisco SIP configuration profile for a particular Cisco IP Phone 7912 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the **create profile command.**
gk<mac>.txt	ASCII text file of a Cisco SIP configuration profile for a particular Cisco IP Phone 7912 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the **file text command.**

Field	Description
Id<mac-address>	Cisco SIP configuration profile for a particular Cisco IP Phone 7905 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the **create profile command.**
Id<mac-address>.txt	ASCII text file of a Cisco SIP configuration profile for a particular Cisco IP Phone 7905 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the **file text command.**
SIPDefault.cnf	Configuration file to be shared by all Cisco SIP IP Phone 7940s and Cisco SIP IP Phone 7960s. This file is automatically generated by the router through the **source-address** and is placed in router memory. The SIPDefault.cnf file contains the IP address that the phones use to register for service, using the Session Initiation Protocol (SIP).
SIP<mac-address>.cnf	Cisco SIP configuration profile for a particular Cisco IP Phone 7940 or Cisco IP Phone 7960 as indicated by the <mac-address>. This file is generated by using the **create profile command.**
syncinfo.xml	Configuration file to be shared by all Cisco SIP IP Phone 7940s and Cisco SIP IP Phone 7960s. This file is generated by using the **create profile command.**

Related Commands

	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phones.
reset (voice register dn)	Performs a complete reboot of one phone associated with a Cisco CME router.
reset (voice register pool)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
text file (voice register global)	Generates an ASCII format text file of the Cisco SIP configuration profile for Cisco IP Phone 7905s and 7905Gs, Cisco IP phone 7912s and 7912Gs, Cisco ATA-186s, and Cisco ATA-188s.
tftp-path (voice register global)	Specifies the directory to which the provisioning file for SIP phones in a Cisco CallManager Express (Cisco CME) system will be written.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

shutdown(telephony-service)

To shut down the Skinny Client Control Protocol (SCCP) server listening socket, use the **shutdown** command in telephony-service configuration mode. To enable service, use the **no** form of this command.

shutdown
no shutdown

Syntax Description

This command has no arguments or keywords.

Command Default

No shutdown is enabled

Command Modes

Telephony-service configuration (config-telephony)
Group configuration (conf-tele-group)
Call-manager-fallback configuration (config-ccm-fallback)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced

Usage Guidelines

The shutdown command allows you to shut down the SCCP server listening sockets when you want to change or remove the IP address set up on your system. For example, If you have IPv6 address and you want to change the IP address set up to dual stack (IPv4 and IPv6) you can use the shutdown command.

Examples

The following example shows SCCP server listening sockets being shut down under telephony-service.

```
Router(config-telephony)#shut down
shutdown
```

The following example shows SCCP server listening sockets being shut down for group 2 (under group mode) in telephony service.

```
Router(config-telephony)#group 2
Router(conf-tele-group)#shutdown
```

The following example shows SCCP server listening sockets being shut down under call-manager-fallback mode.

```
Router(config-telephony)#group 2
```

Router(conf-cm-fallback)#shutdown

Related Commands

Command	Description
protocol -mode	Allows you to configure a preferred IP address mode for SCCP IP phones.
ip source address	Identifies the IP address and port through which IP phones communicate with a Cisco Unified CME router.

sip-prefix

To add "SIP_" prefix in the locale names while populating the configuration files for this phone type in fast track mode, use the **sip-prefix** command in global configuration mode. To disable, use the **no** form of this command.

sip-prefix
no sip-prefix

Syntax Description

This command has no arguments or keywords.

Command Default

sip-prefix is enabled.

Command Modes

Router (config-register-pooltype) #sip-prefix

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced

Usage Guidelines

The sip-prefix command allows you to add “SIP_” prefix in the locale names of the configuration files. For example, DX650 phones do not require “SIP_” prefix in their locale names. However, other phone models require “SIP_” prefix in locale names. Use this command while adding new phones in fast-track mode based on their locale file format.

Examples

The following example shows how to configure “SIP_” prefix on the endpoint Cisco Unified IP Phone 7811.

```
Router(config)#voice register pool-type 7811
Router(config-register-pooltype)#sip-prefix
```

Related Commands

Command	Description
num -lines	Defines number of lines supported by the phone.
phoneload -support	Defines the phone support for Phoneload.
reference -pooltype	Reference pooltype to inherit the properties used in fast-track configuration.

snr

To enable Single Number Reach (SNR) on an extension of an SCCP IP phone, use the **snr** command in ephone-dn configuration mode. To disable SNR on the extension, use the **no** form of this command.

snr *e164-number* **delay** *seconds* **timeout** *seconds* [**cfwd-noan** *extension-number*]
no snr

Syntax Description

e164-number	E.164 telephone number to ring if IP phone extension does not answer.
delay *seconds*	Sets the number of seconds that the call rings the IP phone before ringing the remote phone. Range: 0 to 10. Default: disabled.
timeout *seconds*	Sets the number of seconds that the call rings after the configured delay. Call continues to ring for this length of time on the IP phone even if the remote phone answers the call. Range: 5 to 60. Default: disabled.
cfwd-noan *extension-number*	(Optional) Forwards the call to this target number if the phone does not answer after both the **delay** and **timeout** seconds have expired. This is typically the voice mail number. **Note** This option is not supported for calls from FXO trunks because the calls connect immediately.

Command Default

Single Number Reach is not enabled on the extension.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco UnifiedCME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command enables the SNR feature on the extension. The SNR feature allows users to answer incoming calls on their desktop IP phone or at a remote destination and to pick up in-progress calls on the desktop phone or the remote destination without losing the connection. If an incoming call to this extension is answered immediately, the call is treated as a normal call.

If the call is not answered within the number of seconds set with the **delay** keyword, Cisco Unified CME rings the remote number while continuing to ring the SNR extension. If the call is answered by the desktop IP phone within the number of seconds set with the **timeout** keyword, the call to the remote number is disconnected. If the call is answered on the IP phone, the user can send the call to the remote phone by pressing the Mobility soft key.

If the call is not answered by the IP phone within the number of seconds set with the **timeout** keyword, the ringing call appearance on the IP phone is deleted. This call is marked as hold state on the IP phone. If the user answers the call on the remote phone, the user can pull back the call to the IP phone by pressing the Resume soft-key.

Examples

The following example shows extension 1001 is enabled for SNR. After a call rings at this number for 5 seconds, the call also rings at the remote number 4085550133. The call continues ringing on both phones for 15 seconds. If the call is not answered after a total of 20 seconds, the call no longer rings and is forwarded to the voice-mail number 2001.

```
ephone-dn 10
 number 1001
 mobility
 snr 4085550133 delay 5 timeout 15 cfwd-noan 2001
```

Related Commands

Command	Description
mobility	Enables the Mobility feature on an extension of an SCCP IP phone.
number	Associates a telephone or extension number with an ephone-dn.
softkeys connected	Modifies the order and type of soft keys that display on an IP phone during the connected call state.
softkeys idle	Modifies the order and type of soft keys that display on an IP phone during the idle call state.

snr (voice register dn)

To enable the Single Number Reach (SNR) feature on an extension of a Cisco Unified SIP IP phone, use the **snr** command in voice register dn configuration mode. To disable the SNR feature on the extension, use the **no** form of the command.

snr *e164-number* **delay** *seconds* **timeout** *seconds* [**cfwd-noan** *extension-number*]
no snr

Syntax Description

e164-number	E.164 telephone number to call when the Cisco Unified SIP IP phone extension does not answer.
delay *seconds*	Sets the number of seconds that the Cisco Unified SIP IP phone rings when called. When the time delay is reached, the call is tranferred to the PSTN phone and the SNR directory number. Range: 0 to 30. Default: 5.
timeout *seconds*	Sets the number of seconds that the Cisco Unified SIP IP phone rings after the configured time delay. When the timeout value is reached, no call is displayed on the phone. You have to use the Resume soft key to pull back or the Mobility soft key to send the call to a mobile phone. Range: 30 to 60. Default: 60. **Note** When the default is enabled, the Cisco Unified SIP IP phone continues to ring for 60 seconds even if the remote phone answers the call.
cfwd-noan *extension-number*	(Optional) Forwards the call to the extension number when the phone does not answer after both the time delay and timeout values are reached. The extension number is typically the voice mail number. **Note** This option is not supported for calls from FXO trunks because the calls connect immediately.

Command Default

The SNR feature is not enabled on the extension of a Cisco Unified SIP IP phone.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **snr** command to enable the SNR feature on an extension of a Cisco Unified SIP IP phone.

The SNR feature allows you to answer incoming calls on your desktop IP phones or at a remote destination. It also allows you to pick up in-progress calls on a desktop phone or at a remote destination without losing the connection. If an incoming call to the extension is answered immediately, the call is treated as a normal call.

If the call is not answered within the number of seconds set with the **delay** keyword, Cisco Unified CME rings the remote number while continuing to ring the SNR extension. If the call is answered by the desktop IP phone within the number of seconds set with the **timeout** keyword, the call to the remote number is disconnected. If the call is answered on the IP phone, you can send the call to the remote phone by pressing the Mobility soft key.

If the call is not answered by the IP phone within the number of seconds set with the **timeout** keyword, the call is dislayed on the IP phone as being in the hold state. If the user answers the call on the remote phone, the user can pull back the call to the IP phone by pressing the Resume soft key.

Examples

The following example shows that extension 1004 is enabled for SNR. After a call rings at this number for one second, the call also rings at the remote number 9900. The call continues ringing on both phones for 10 seconds. If the call is not answered after a total of 11 seconds, the call no longer rings and is forwarded to the voice-mail number 1007.

```
Router(config)# voice register dn  3
Router(config-register-dn)# number 1004
Router(config-register-dn)# name John Smith
Router(config-register-dn)# mobility
Router(config-register-dn)# snr calling-number local
Router(config-register-dn)# snr 9900 delay 1 timeout 10 cfwd-noan 1007
Router(config-register-dn)# snr ring-stop
Router(config-register-dn)# snr answer-too-soon 2
```

Related Commands

Command	Description
mobility (voice register dn)	Enables the Mobility feature on an extension of a Cisco Unified SIP IP phone.
snr answer-too-soon (voice register dn)	Sets the time in which SNR calls are prevented from being diverted to the voice mailbox of a mobile phone.
snr calling-number local (voice register dn)	Replaces the calling party number displayed on the configured mobile phone with the local SNR number.
snr ring-stop (voice register dn)	Ends the ringing on a Cisco Unified SIP IP phone after the Single SNR call is answered on the configured mobile phone.

snr answer-too-soon

To set the SNR answer to soon timer, use the snr answer-too-soon command in ephone-dn mode. To reset the default, use the no form of the command.

snr answer-too-soon time
no snr answer-too-soon

Syntax Description

time	Time, in seconds. Range: 1 to 5.

Command Default

No answer too soon timer is set.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to enable timer for answering the call on an SNR mobile phone. You can set a timer from 1 to 5 seconds. If the call is answered within the timer, the mobile leg is disconnected.

Examples

```
Router(config)ephone-dn 10
Router(config-ephone-dn)#snr answer-too-soon 4
```

Related Commands

Command	Description
snr	**Enables SNR on the extension of an SCCP IP phone.**

snr answer-too-soon (voice register dn)

To set the time in which Single Number Reach (SNR) calls are prevented from being diverted to the voice mailbox of a mobile phone, use the **snr answer-too-soon** command in voice register dn configuration mode. To allow SNR calls to be diverted to the voice mailbox, use the **no** form of the command.

snr answer-too-soon *time*
no snr answer-too-soon

Syntax Description

time	Time, in seconds. Range: 1 to 5.

Command Default

No answer-too-soon time is set. Calls may be diverted to the voice mailbox when a user's mobile phone is not answered or is turned off.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

Use the **snr answer-too-soon** command to set the time in which SNR calls are prevented from being diverted to the voice mailbox of a mobile phone. When the call is diverted to the voice mailbox within the set time, the mobile phone call leg is disconnected.

Examples

The following example shows how SNR calls are prevented from being diverted to the voice mailbox of a mobile phone for 2 seconds:

```
Router(config)# voice register dn  3
Router(config-register-dn)# number 1004
Router(config-register-dn)# name John Smith
Router(config-register-dn)# mobility
Router(config-register-dn)# snr calling-number local
Router(config-register-dn)# snr 9900 delay 1 timeout 10
Router(config-register-dn)# snr ring-stop
Router(config-register-dn)# snr answer-too-soon 2
```

Related Commands

Command	Description
snr (voice register dn)	Enables the SNR feature on an extension of a Cisco Unified SIP IP phone.

snr calling-number local

To replace the calling-party number with the single number reach (SNR) extension number in calls forwarded to the remote phone, use the **snr calling-number local** command in ephone-dn configuration mode. To reset to the default, use the **no** form of this command.

snr calling-number local
no snr calling-number local

Syntax Description

This command has no arguments or keywords.

Command Default

Calling-party number is not replaced.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command replaces the original calling party number with the SNR extension number (local number) in the caller ID display for SNR calls forwarded to the remote phone. When the call is forwarded to the remote phone, such as a mobile phone, the caller ID shows the SNR number that the caller dialed, not the number of the original calling party.

Examples

The following example shows that the original calling party number is replaced by the SNR extension number 1234 when the call is forwarded to the mobile phone:

```
ephone-dn 1
 number 1234
 mobility
 snr 4085550123 delay 5 timeout 15 cfwd-noan 2001
 snr calling-number local
```

Related Commands

Command	Description
calling-number local	Replaces a calling-party number and name with the forwarding-party number and name for all calls.
mobility	Enables the Mobility feature on an extension of an SCCP IP phone.
snr	Enables SNR on the extension of an SCCP IP phone.

snr calling-number local (voice register dn)

To replace the calling party number displayed on the configured mobile phone with the local Single Number Reach (SNR) number, use the **snr calling-number local** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

snr calling-number local
no snr calling-number local

Syntax Description

This command has no arguments or keywords.

Command Default

The number of the calling party is displayed on the mobile phone configured to receive SNR calls.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Examples

The following example shows how the **snr calling-number local** command is used to display the local SNR number instead of the calling party's number on the mobile phone:

```
Router(config)# voice register dn  3
Router(config-register-dn)# number 1004
Router(config-register-dn)# name John Smith
Router(config-register-dn)# mobility
Router(config-register-dn)# snr calling-number local
Router(config-register-dn)# snr 9900 delay 1 timeout 10
Router(config-register-dn)# snr ring-stop
Router(config-register-dn)# snr answer-too-soon 2
```

Related Commands

Command	Description
snr (voice register dn)	Enables the SNR feature on an extension of a Cisco Unified SIP IP phone.

snr mode

To set the mode for the Single Number Reach (SNR) directory number (DN), use the **snr mode** command in ephone-dn configuration mode. To return to the default, use the **no** form of this command.

snr mode [**virtual**]
no snr mode

Syntax Description

virtual	Enables the virtual mode for an SNR DN when it is unregistered or floating. **Note** Virtual mode is activated when the DN state remains up when it should be in the down state.

Command Default

No DN mode is set for the SNR feature.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

A virtual SNR DN is a DN not associated with any registered phone but is a number that can be called, have its calls forwarded to a preconfigured mobile phone, or put on an Auto Hold state when the mobile phone answers the call or the time delay is reached. In the Auto Hold state, the DN can either be floating or unregistered. A floating DN is a DN not configured with any phone while an unregistered DN is one associated with phones not registered to a Cisco Unified CME system.

A ringback tone is heard when a call is made to a virtual DN.

To enable the SNR feature, the SNR DN must be in the up state, the Mobility feature must be enabled, and the time delay or timeout value configured with the **snr** command must be reached.

Examples

The following example sets the virtual DN mode for SNR on ephone-dn 1:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# snr mode virtual
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode to configure a DN for an IP phone line, intercom line, paging line, voice-mail port, or MWI.
snr	Enables SNR on an extension of a Cisco Unified SCCP IP phone.

snr ring-stop

To stop the IP phone from ringing after the SNR call is answered on a mobile phone, use the snr ring-stop command in ephone-dn configuration mode. To reset the default value, use the no form of the command.

snr ring-stop
no snr ring-stop

Syntax Description

This command has no arguments or keywords.

Command Default

Phone continues to ring after the SNR call is answered on a mobile phone.

Command Modes

Ephone-dn configuration (conf-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to stop the IP phone from ringing after the SNR call is answered on a mobile phone.

Examples

```
Router(config-ephone-dn)10
Router(config-ephone-dn)#snr ring-stop
```

Related Commands

Command	Description
snr	**Enables SNR on the extension of an SCCP IP phone.**

snr ring-stop (voice register dn)

To end the ringing on a Cisco Unified SIP IP phone after the Single Number Reach (SNR) call is answered on the configured mobile phone, use the **snr ring-stop** command in voice register dn configuration mode. To allow the Cisco Unified SIP IP phone to continue ringing even after the SNR call has been answered, use the **no** form of the command.

snr ring-stop
no snr ring-stop

Syntax Description

This command has no arguments or keywords.

Command Default

The Cisco Unified SIP IP phone continues to ring even after the SNR call is answered on a mobile phone.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Examples

The following example shows how to end the ringing on a Cisco Unified SIP IP phone:

```
Router(config)# voice register dn  3
Router(config-register-dn)# number 1004
Router(config-register-dn)# name John Smith
Router(config-register-dn)# mobility
Router(config-register-dn)# snr calling-number local
Router(config-register-dn)# snr 9900 delay 1 timeout 10
Router(config-register-dn)# snr ring-stop
Router(config-register-dn)# snr answer-too-soon 2
```

Related Commands

Command	Description
snr (voice register dn)	Enables the SNR feature on an extension of a Cisco Unified SIP IP phone.

softkeys alerting

To configure an ephone template for soft-key display during the alerting call stage, use the **softkeys alerting** command in ephone-template configuration mode. To remove a **soft key alerting** configuration, use the **no** form of this command.

softkeys alerting [**Acct**] [**Callback**] [**Endcall**]
no softkeys alerting

Syntax Description

Acct	(Optional) Soft-key name that appears on the IP phone during the alerting call stage. Short for "account code." Provides access to configured accounts.
Callback	(Optional) Soft-key name that appears on the IP phone during the alerting call stage. Requests callback notification when a busy called line becomes free.
Endcall	(Optional) Soft-key name that appears on the IP phone during the alerting call stage. Ends the current call.

Command Default

The default soft keys for the alerting call stage and the order in which they appear on IP phones are, from first to last, Acct, Callback, and Endcall.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	3.2	This command was introduced.

Usage Guidelines

The alerting call stage is when the remote point is being notified of an incoming call, and the status of the remote point is being relayed to the caller as either ringback or busy.

The number and order of soft keys listed in the **softkeys alerting** correspond to the number and order of soft keys that will appear on IP phones.

Examples

In the following example, ephone template 1 is configured for the alerting stage and for the seized and connected call stages:

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1
Router(config-ephone-template)# softkeys seized Redial Cfwdall Pickup

Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an ephone.
softkeys connected	Configures an ephone template for soft-key display during the connected call stage.

Command	Description
softkeys idle	Configures an ephone template for soft-key display during the idle call stage.
softkeys seized	Configures an ephone template for soft-key display during the seized call stage.

softkeys connected (voice register template)

To modify the soft key display during the connected call state on Cisco Unified SIP IP phones, use the **softkeys connected** command in voice register template configuration mode. To return to the default, use the **no** form of this command.

softkeys connected [**ConfList**] [**Confrn**] [**Endcall**] [**Hold**] [**Park**] [**RmLstC**] [**Trnsfer**] [**iDivert**] [**HLog**]
no softkeys connected

Syntax Description

ConfList	(Optional) Lists all the participants in a conference.
Confrn	(Optional) Connects callers to a conference call. This soft key also enables ad-hoc conference creators to initiate a conference.
Endcall	(Optional) Ends the current call.
Hold	(Optional) Places an active call on hold and resumes the call.
Park	(Optional) Places an active call on hold so it can be retrieved from another phone in the system.
RmLstC	(Optional) Removes the last conference participant.
Trnsfer	(Optional) Transfers active calls to another extension.
iDivert	(Optional) Immediately diverts a call to a voice-messaging system.
HLog	(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the **hunt-group logout** command to HLog for this softkey to be functional. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.

Command Default

The default soft keys for the connected call state and the order in which they appear on Cisco Unified SIP IP phones are, from first to last: ConfList, Confrn, Endcall, Hold, Park, RmLstC, Trnsfer, iDivert, and HLog.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(22)YB	Cisco Unified CME 7.1	The **Park** keyword was added.
15.1(3)T	Cisco Unified CME 8.5	The **iDivert** keword was added.
15.2(2)T	Cisco Unified CME 9.0	This command was modified. The syntax description for the Confrn soft key was updated. The **ConfList** and **RmLastC** keywords were added.

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1 15.6(3)M1	Cisco Unified CME 11.6	HLog Softkey support was introduced.

Usage Guidelines

The connected call state is when the connection to a remote point is established.

The number and order of soft keys used in this command correspond to the number and order of soft keys that appear on Cisco Unified SIP IP phones. Any soft key that is not explicitly specified with this command is disabled.

The ConfList and RmLastC soft keys are added in the connected state when hardware conference is enabled.

This command is not supported on the Cisco Unified 7905, 7912, 7940, and 7960 SIP IP Phones.

Examples

In the following example, Cisco Unified SIP IP phone template 1 is configured for the connected and seized call states:

```
Router(config)# voice register template 1
Router(config-register-temp)# softkeys seized Redial Cfwdall EndCall HLog

Router(config-register-temp)# softkeys connected Confrn Hold Endcall HLog
```

The following is a sample output from the **show voice register template** command. The output shows that the iDivert soft key is in connected state.

```
Router# show voice register template 1
Temp Tag 1
Config:
  Attended Transfer is enabled
  Blind Transfer is enabled
  Semi-attended Transfer is enabled
  Conference is enabled
  Caller-ID block is disabled
  DnD control is enabled
  Anonymous call block is disabled
  softkeys seized Redial Cfwdall EndCall HLog
  softkeys connected Confrn Hold Endcall HLog
```

Related Commands

Command	Description
softkeys hold (voice register template)	Modifies the soft key display on Cisco Unified SIP IP phones during the hold call state.
softkeys idle (voice register template)	Modifies the soft key display on Cisco Unified SIP IP phones during the idle call state.
softkeys seized (voice register template)	Modifies the soft key display on Cisco Unified SIP IP phones during the seized call state.
template (voice register pool)	Applies a phone template to a Cisco Unified SIP IP phone.

softkeys connected

To modify the order and type of soft keys that display on an IP phone during the connected call state, use the **softkeys connected** command in ephone-template configuration mode. To reset to the default, use the **no** form of this command.

softkeys connected [**Acct**] [**ConfList**] [**Confrn**] [**Endcall**] [**Flash**] [**HLog**] [**Hold**] [**Join**] [**LiveRcd**] [**Mobility**] [**Park**] [**RmLstC**] [**Select**] [**TrnsfVM**] [**Trnsfer**]
no softkeys connected

Syntax Description

Acct	(Optional) Soft key that provides access to configured accounts.
ConfList	(Optional) Soft key that lists all parties in a conference.
Confrn	(Optional) Soft key that connects callers to a conference call.
Endcall	(Optional) Soft key that ends the current call.
Flash	(Optional) Soft key that provides hookflash functionality for public switched telephony network (PSTN) services on calls connected to the PSTN via a foreign exchange office (FXO) port. Also called “hookflash.”
HLog	(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the **hunt-group logout** command to HLog for this soft key to be visible. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.
Hold	(Optional) Soft key that places an active call on hold and resumes the call.
Join	(Optional) Soft key that joins an established call to conference.
LiveRcd	(Optional) Soft key that enables recording of a call.
Mobility	(Optional) Soft key that forwards the call to the PSTN number defined by the Single Number Reach (SNR) feature.
Park	(Optional) Soft key that places an active call on hold, so it can be retrieved from another phone in the system.
RmLstC	(Optional) Soft key that removes the last party added to the conference. This soft key only works for the conference creator.
Select	(Optional) Soft key that selects a call or a conference on which to take action.
TrnsfVM	(Optional) Soft key that transfers a call to a voice-mail extension number.
Trnsfer	(Optional) Soft key that transfers active calls to another extension.

Command Default

The default soft keys for the connected call state and the order in which they appear on IP phones are, from first to last:

- With HLog support: Hold, EndCall, Trnsfer, Confrn, Acct, Flash, Park, HLog

- Without HLog support: Hold, EndCall, Trnsfer, Confrn, Acct, Flash, Park

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **HLog** keyword was added.
12.4(11)XJ	Cisco Unified CME 4.1	The **ConfList**, **Join**, **RmLstC**, and **Select** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(15)XZ	Cisco Unified CME 4.3	The **LiveRcd** and **TrnsfVM** keywords were added.
12.4(20)T	Cisco Unified CME 7.0	This command with the **LiveRcd** and **TrnsfVM** keywords was integrated into Cisco IOS Release 12.4(20)T.
12.4(22)YB	Cisco Unified CME 7.1	The **Mobility** keyword was added.
12.4(24)T	Cisco UnifiedCME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

The connected call state is when the connection to a remote point has been established.

Configure the ConfList, Join, and RmLstC soft keys for conferencing functions. These soft keys are supported for hardware-based conferencing only and require the appropriate DSP farm configuration.

Note The ConfList (including the Remove, Update, and Exit soft keys within the ConfList function) and RmLstC soft keys do not work on the Cisco Unified IP Phone 7902, 7935, and 7936.

Examples

In the following example, ephone template 1 modifies the soft keys displayed for the seized, alerting, and connected call states:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# softkeys seized Redial Cfwdall Pickup

Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone.
ephone-template (ephone)	Applies an ephone template to an ephone.
fxo-hook-flash	Enables display of the Flash soft key.

Command	Description
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of the HLog soft key on phones.
softkeys alerting	Modifies the soft-key display for the alerting call state.
softkeys idle	Modifies the soft-key display for the idle call state.
softkeys ringing	Modifies the soft-key display for the ringing call state.
softkeys seized	Modifies the soft-key display for the seized call state.

softkeys hold

To configure an ephone template to modify soft-key display during the call-hold call stage, use the **softkeys hold** command in ephone-template configuration mode. To remove a **softkeys hold** configuration, use the **no** form of this command.

softkeys hold [**Join**] [**Newcall**] [**Resume**] [**Select**]
no softkeys hold

Syntax Description

Join	(Optional) Soft-key name that appears on an IP phone during the hold call stage. Joins an established call to a conference.
Newcall	(Optional) Soft-key name that appears on an IP phone during the hold call stage. Opens a line on a speaker phone to place a new call.
Resume	(Optional) Soft-key name that appears on an IP phone during the hold call stage. Reconnects with the call on hold.
Select	(Optional) Soft-key name that appears on an IP phone during the hold call stage. Selects a call or a conference on which to take action.

Command Default

The default soft keys for the hold call stage and the order in which they appear on IP phones are alphabetical, from first to last, Join, Newcall, Resume, and Select.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **Join** and **Select** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	This command with the **Join** and **Select** keywords was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

You reach the call-hold state by pressing the Hold soft key while you are in the connected state. From the hold state, you can press Resume to return to the connected state or NewCall to start another call, leaving the original call in the call-hold state.

The number and order of soft keys listed in the **softkeys hold** correspond to the number and order of soft keys that will appear on IP phones.

Configure the Join and Select soft keys for conferencing functions. These soft keys are supported for hardware-based conferencing only and require the appropriate DSP farm configuration..

Examples

In the following example, ephone template 1 is configured for the idle, alerting, connected, and hold call stages. It is applied to ephone 25. When ephone 25 has a call on hold, the only soft key that will be available is the Resume soft key.

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1

Router(config-ephone-template)# softkeys idle Redial Cfwdall Pickup

Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
Router(config-ephone-template)# softkeys hold Resume
Router(config-ephone-template)# exit
Router(config)# ephone 25
Router(config-ephone)# button 1:39
Router(config-ephone)# ephone-template 1
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone.
ephone-template	Declares and names an ephone template to configure IP phone soft-key display and enters ephone-template configuration mode
ephone-template (ephone)	Applies an ephone template to an ephone.
softkeys alerting	Configures an ephone template for soft-key display during the alerting call stage.
softkeys connected	Configures an ephone template for soft-key display during the connected call stage.
softkeys idle	Configures an ephone template for soft-key display during the idle call stage.
softkeys seized	Configures an ephone template for soft-key display during the seized call stage.

softkeys idle

To modify the order and type of soft keys that display on an IP phone during the idle call state, use the **softkeys idle** command in ephone template configuration mode. To reset to the default, use the **no** form of this command.

softkeys idle [**Cfwdall**] [**ConfList**] [**Dnd**] [**Gpickup**] [**HLog**] [**Join**] [**Login**] [**Mobility**] [**Newcall**] [**Pickup**] [**Redial**] [**RmLstC**]
no softkeys idle

Syntax Description

Cfwdall	(Optional) Soft key that forwards all calls.
ConfList	(Optional) Soft key that lists all parties in a conference.
Dnd	(Optional) Soft key that enables the Do-Not-Disturb features. This key is a toggle; pressing it a second time disables DND.
Gpickup	(Optional) Soft key that selectively picks up calls coming into a phone number that is a member of a pickup group.
HLog	(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the **hunt-group logout** command to HLog for this soft key to be visible. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.
Join	(Optional) Soft key that joins an established call to a conference.
Login	(Optional) Soft key that provides personal identification number (PIN)-controlled access to restricted phone features.
Mobility	(Optional) Soft key that enables Single Number Reach (SNR) feature. This key is a toggle; pressing it a second time disables SNR.
Newcall	(Optional) Soft key that opens a line on a speaker phone to place a new call.
Pickup	(Optional) Soft key that selectively picks up calls coming into another extension.
Redial	(Optional) Soft key that redials the last number dialed.
RmLstC	(Optional) Soft key that removes the last party added to the conference. This soft key removes the last party only when the conference creator presses it.

Command Default

The default soft keys for the idle call stage and the order in which they appear on IP phones are:

- FXO Trunk: Redial, NewCall, DoNotDisturb
- With HLog support: Redial, NewCall, CFwdAll, CallPickUp, GrpCallPickUp, DoNotDisturb, Login, HLog
- Without HLog support: Redial, NewCall, CFwdAll, CallPickUp, GrpCallPickUp, DoNotDisturb, Login

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **HLog** keyword was added.
12.4(9)T	Cisco Unified CME 4.0	The **HLog** keyword was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **ConfList**, **Join**, and **RmLstC** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	This command with the **ConfList**, **Join**, and **RmLstC** keywords was integrated into Cisco IOS Release 12.4(15)T.
12.4(22)YB	Cisco Unified CME 7.1	The **Mobility** keyword was added.
12.4(24)T	Cisco unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

The idle calling stage occurs before a call is made and after a call is complete.

The number and order of soft keys listed in the **softkeys idle** command correspond to the number and order of soft keys on IP phones.

Configure the ConfList, Join, and RmLstC soft keys for conferencing functions. These soft keys are supported for hardware-based conferencing only and require the appropriate DSP farm configuration.

Note The ConfList (including the Remove, Update, and Exit soft keys within the ConfList function) and RmLstC soft keys do not work on the Cisco Unified IP Phone 7902 and Cisco Unified IP Phone 7935 and 7936.

Examples

In the following example, ephone template 1 is configured for the idle stage and for the alerting and connected call stages:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# softkeys idle Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone.
ephone-template	Creates an ephone template.
ephone-template (ephone)	Applies an ephone template to an ephone.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.
softkeys alerting	Configures soft-key display during the alerting call state.

Command	Description
softkeys connected	Configures soft-key display during the connected call state.
softkeys seized	Configures soft-key display during the seized call state.

softkeys idle (voice register template)

To modify the soft-key display during the idle call state on SIP phones, use the **softkeys idle** command in voice register template configuration mode. To remove a **softkeys idle** configuration, use the **no** form of this command.

softkeys idle [**Cfwdall**] [**DND**] [**Gpickup**] [**Newcall**] [**Pickup**] [**Redial**] [**HLog**]
no softkeys idle

Syntax Description

Cfwdall	(Optional) Soft key for "call forward all." Forwards all calls.
DND	(Optional) Soft key that enables the Do-Not-Disturb feature.
Gpickup	(Optional) Soft key that allows a user to pickup a call that is ringing on another phone.
Newcall	(Optional) Soft key that opens a line on a speakerphone to place a new call.
Pickup	(Optional) Soft key that allows a user to pickup a call that is ringing on another phone that is a member of the same pickup group.
Redial	(Optional) Soft key that redials the last number dialed.
HLog	(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the **hunt-group logout** command to HLog for this softkey to be functional. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.

Command Default

The default soft keys for the idle call state and the order in which they appear on SIP phones are, from first to last, Redial, Newcall, Cfwdall, and HLog.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(22)YB	Cisco Unified CME 7.1	The **DND** keyword was added.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.
Cisco IOS XE Everest 16.4.1 15.6(3)M1	Cisco Unified CME 11.6	HLog Softkey support was introduced.

Usage Guidelines

The idle calling state occurs before a call is made and after a call is complete.

The number and order of soft keys used in this command correspond to the number and order of soft keys that appear on SIP phones. Any soft key that is not explicitly specified with this command is disabled if this command is used to change the default soft keys.

This command is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

Examples

In the following example, SIP phone template 1 is configured for the idle and connected call states:

```
Router(config)# voice register template 1
Router(config-register-template)# softkeys idle Redial Cfwdall HLog
Router(config-register-template)# softkeys connected Confrn Hold Endcall HLog
```

Related Commands

Command	Description
softkeys connected (voice register template)	Modifies the soft-key display on SIP phones during the connected call state.
softkeys hold (voice register template)	Modifies the soft-key display on SIP phones during the hold call state.
softkeys idle (voice register template)	Modifies the soft-key display on SIP phones during the idle call state.
softkeys seized (voice register template)	Modifies the soft-key display on SIP phones during the seized call state.
template (voice register pool)	Applies a phone template to a SIP phone.

softkeys personal-conf-user (voice register template)

To enable a personal user softkey template for Cisco IP Conference Phones 7832 and 8832, use the **softkeys personal-conf-user** command in voice register template configuration mode. To switch to a public user softkey template, use the **no** form of this command.

softkeys personal-conf-user
no softkeys personal-conf-user

Command Default

By default, the CLI command **softkeys personal-conf-user** is disabled. Hence, the Cisco IP Conference Phones 7832 and 8832 support the public user softkey template if the command is not configured.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Fuji 16.9.1	Unified CME 12.3	A personal and public softkey template support was introduced for new Softkeys introduced as part of Unified CME 12.3 Release for Cisco IP Conference Phone 7832 and 8832.

Usage Guidelines

The CLI command **softkeys personal-conf-user** is an optional configuration that is required only when the phone template is applied to Cisco IP Conference Phones 7832 and 8832. If no configuration is provided, then the default configuration of public user softkey template is applied. When the CLI command is enabled, the personal softkey template is applied to the conference phone (Only for Cisco IP Conference Phone 7832 and 8832). When the command is not enabled, the public softkey template is applied to the conference phone (Only for Cisco IP Conference Phone 7832 and 8832).

As compared to public softkey user template, the following softkeys are additionally supported in a personal user softkey template for various phone states:

- Messages
- CfwdAll
- DND
- Redial

Examples

In the following example, Cisco IP Conference Phone 7832 is configured for the personal softkeys template:

```
Router(config)# voice register template 7
Router(config-register-template)# softkeys personal-conf-user
```

Related Commands

Command	Description
softkeys connected (voice register template)	Modifies the soft-key display on SIP phones during the connected call state.

Command	Description
softkeys hold (voice register template)	Modifies the soft-key display on SIP phones during the hold call state.
softkeys idle (voice register template)	Modifies the soft-key display on SIP phones during the idle call state.
softkeys seized (voice register template)	Modifies the soft-key display on SIP phones during the seized call state.
template (voice register pool)	Applies a phone template to a SIP phone.

softkeys remote-in-use

To modify the order and type of soft keys that display on the IP phone during the remote-in-use call state, use the **softkeys remote-in-use** command in ephone-template configuration mode. To reset to the default, use the **no** form of this command.

softkeys remote-in-use [**CBarge**] [**Newcall**]
no softkeys remote-in-use

Syntax Description

CBarge	(Optional) Soft key that allows a user to barge into a call on a shared octo-line directory number.
Newcall	(Optional) Soft key that opens a line on a speakerphone to place a new call.

Command Default

The default soft keys for the remote-in-use call state and the order in which they appear on IP phones are Newcall, CBarge.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was itegrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

The remote-in-use call state is when another phone is connected to a call on an octo-line directory number shared by this phone.

Examples

In the following example, ephone template 1 modifies the soft keys displayed for the alerting, connected, and remote-in-use call states:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
Router(config-ephone-template)# softkeys remote-in-use CBarge Newcall
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an ephone.
softkeys alerting	Modifies the soft-key display for the alerting call stage.
softkeys idle	Modifies the soft-key display for the idle call stage.
softkeys seized	Modifies the soft-key display for the seized call stage.

softkeys remote-in-use (voice register template)

To modify the soft-key display during the remote-in-use call state on SIP shared-line phones, use the **softkeys remote-in-use** command in voice register template configuration mode. To reset to the default, use the **no** form of this command.

softkeys remote-in-use [**Barge**] [**Newcall**] [**cBarge**]
no softkeys remote-in-use

Syntax Description

Barge	(Optional) Soft key that allows a user to join a call on a shared line.
Newcall	(Optional) Soft key that opens a line on a phone to place a new call.
cBarge	(Optional) Soft key that allows a user to join a call on a shared line and to turn the call into a conference call.

Command Default

The default soft keys for the remote-in-use call state and the order in which they appear on SIP phones are Barge, Newcall, cBarge.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

The remote-in-use call state is when another phone is connected to a call on a directory number shared by this phone.

Examples

In the following example, SIP phone template 1 modifies the soft keys displayed for the alerting, connected, and remote-in-use call states:

```
Router(config)# voice register template 1
Router(config-register-temp)# softkeys alerting Callback Endcall
Router(config-register-temp)# softkeys connected Confrn Hold Endcall
Router(config-register-temp)# softkeys remote-in-use CBarge Newcall
```

Related Commands

Command	Description
softkeys alerting (voice register template)	Modifies the soft-key display on SIP phones during the alerting call state.
softkeys idle (voice register template)	Modifies the soft-key display on SIP phones during the idle call state.

Command	Description
softkeys seized (voice register template)	Modifies the soft-key display on SIP phones during the seized call state.
template (voice register pool)	Applies a phone template to a SIP phone.

softkeys ringin (voice register template)

To modify the soft-key display during the ringing call state on SIP phones, use the **softkeys ringIn** command in voice register template configuration mode. To remove the **softkeys ringIn** configuration, use the **no** form of this command.

softkeys ringIn [**Answer**] [**DND**] [**iDivert**] [**HLog**]
no softkeys ringIn

Syntax Description

Answer	(Optional) Soft key that picks up an incoming call.
DND	(Optional) Soft key that enables the Do Not Disturb feature.
HLog	(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the **hunt-group logout** command to HLog for this soft key to be functional. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.
iDivert	(Optional) Immediately diverts a call to a voice-messaging system.

Command Default

The following soft keys are displayed in alphabetical order, first to last, on IP phones during the ringIn call state: Answer, Dnd, HLog, and iDivert.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.
Cisco IOS XE Everest 16.4.1 15.6(3)M1	Cisco Unified CME 11.6	HLog Softkey support was introduced.

Usage Guidelines

Use this command to create a template in which you specify which soft keys are displayed, and in what order, on an IP phone during the ringing call state. The ringing calling state is after a call is received and before the call is connected.

Any soft key that is not explicitly specified with this command is disabled if this command is used to change the default soft keys.

Configure the **Answer** keyword to enable a phone user to answer an incoming call on a line button that is unavailable; for example, if a line button is configured with a dual-line directory number and a call is holding on one channel of the directory number and another call is ringing on the second channel, the phone user can press the Answer soft key to pick up the incoming call on the second channel.

Configure the **DND** keyword to enable the phone user to place the phone into Do-Not-Disturb mode. Configure the Dnd soft key and the **hunt-group logout DND** command to enable the phone user to invoke DND mode and log the phone out of hunt groups in which it is a member.

Configure the **iDivert** keyword to immediately divert a call to a voice-messaging system.

Configure the **HLog** keyword to place a phone into not-ready status, in which it does not accept hunt-group calls.

To apply an voice register template to a phone, configure the **voice register template** command in voice register pool configuration mode.

Examples

In the following example, SIP phone template 1 is configured for the ringing state, and for the alerting and connected call states:

```
Router(config)# voice register template 1
Router(config-register-template)# softkeys ringIn Answer Dnd Hlog iDivert

Router(config-register-template)# softkeys idle Newcall Redial Pickup Cfwdall HLog
Router(config-register-template)# softkeys connected Transfer Hold Endcall HLog
```

Related Commands

Command	Description
softkeys connected (voice register template)	Modifies the soft-key display on SIP phones during the connected call state.
softkeys hold (voice register template)	Modifies the soft-key display on SIP phones during the hold call state.
softkeys idle (voice register template)	Modifies the soft-key display on SIP phones during the idle call state.
softkeys seized (voice register template)	Modifies the soft-key display on SIP phones during the seized call state.

softkeys ringing

To configure an ephone template for soft-key display during the ringing call state, use the **softkeys ringing** command in ephone-template configuration mode. To remove the **softkeys ringing** configuration, use the **no** form of this command.

softkeys ringing [**Answer**] [**Dnd**] [**HLog**]
no softkeys ringing

Syntax Description

Answer	(Optional) Soft-key name that appears on the IP phone during the ringing call state.
Dnd	(Optional) Soft-key name that appears on the IP phone during the ringing call state.
HLog	(Optional) Soft-key name that appears on the IP phone during the ringing call state.

Command Default

The following soft keys are displayed in alphabetical order, first to last, on IP phones during the ringing call state: Answer, Dnd, HLog

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command in ephone-template configuration mode to create a template in which you can specify which soft keys are displayed, and in what order, on an IP phone during the ringing call state. The ringing calling state is when a call is received and before the call is connected.

Any soft key that is not explicitly configured is disabled.

You can enter any of the keywords in any order. The number and order of soft keys listed in the **softkeys ringing** command corresponds to the number and order of soft keys that will appear on IP phones during the ringing call state.

Configure the **Answer** keyword with this command to enable a phone user to answer an incoming call on a line button that is unavailable; for example, if a line button is configured with a dual-line directory number and a call is holding on one channel of the directory number and another call is ringing on the second channel, the phone user can use the Answer soft key to pick up the incoming call on the second channel.

Configure the **HLog** keyword with this command to *display* the Hlog soft key during the ringing call state. To enable HLog softkey *functionality* during the call ringing state, you must also configure the **hunt-group logout HLog** command. If you configure the Hlog soft key and do not configure the **hunt-group logout HLog** command, the Hlog soft key appears on the phone screen but is not functional. The HLog softkey is a toggle for enabling or disabling the not-ready status, in which the directory number does not accept hunt-group calls.

Configure the **Dnd** keyword with this command to enable the phone user to place the phone into Do-Not-Disturb mode. Configure the Dnd soft key and the **hunt-group logout DND** command to enable the phone user to invoke DND mode and log the phone out of hunt groups in which it is a member.

To apply an ephone template to phone, configure the **ephone-template (ephone)** command in the ephone configuration mode.

Examples

In the following example, ephone template 1 is configured for the ringing state, and for the alerting and connected call states:

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1

Router(config-ephone-template)# softkeys ringing Answer Dnd Hlog

Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
dnd feature ring	Allows phone buttons configured with the feature-ring option to not ring when their phones are in do-not-disturb (DND) mode.
ephone-template (ephone)	Applies an ephone template to an ephone.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents.
softkeys alerting	Configures an ephone template for soft-key display during the alerting call state.
softkeys connected	Configures an ephone template for soft-key display during the connected call state.
softkeys idle	Configures an ephone template for soft-key display during the idle call state.
softkeys seized	Configures an ephone template for the soft-key display during the seized call state.

softkeys seized

To modify the order and type of soft keys that display on an IP phone during the seized call state, use the **softkeys seized** command in ephone-template configuration mode. To remove a **softkeys seized** configuration, use the **no** form of this command.

softkeys seized [**CallBack**] [**Cfwdall**] [**CWOff**] [**Endcall**] [**Gpickup**] [**HLog**] [**MeetMe**] [**Pickup**] [**Redial**]
no softkeys seized

Syntax Description

CallBack	(Optional) Soft key that requests callback notification when a busy called line becomes free.
Cfwdall	(Optional) Soft key that forwards all calls.
CWOff	(Optional) Soft key that disables Call Waiting.
Endcall	(Optional) Soft key that ends the current call.
Gpickup	(Optional) Soft key that selectively picks up calls coming into a phone number that is a member of a pickup group.
HLog	(Optional) Soft key that places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the **hunt-group logout** command to HLog for this soft key to be visible. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.
MeetMe	(Optional) Soft key that initiates a meet-me conference.
Pickup	(Optional) Soft key that selectively picks up calls to another extension.
Redial	(Optional) Soft key that redials the last number dialed.

Command Default

The default soft keys for the seized call stage and the order in which they appear on IP phones are:

- With HLog support: Redial, EndCall, CFwdAll, CallPickUp, GrpCallPickUp, CallBack, HLog
- Without HLog support: Redial, EndCall, CFwdAll, CallPickUp, GrpCallPickUp, CallBack

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **HLog** keyword was added.
12.4(9)T	Cisco Unified CME 4.0	The **HLog** keyword was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **MeetMe** keyword was added.
12.4(15)T	Cisco Unified CME 4.1	The **MeetMe** keyword was integrated into Cisco IOS Release 12.4(15)T.

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. The **CWOff** keyword was added.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

The seized calling stage is when a caller is attempting a call and has not yet been connected.

The number and order of soft keys listed in the **softkeys seized** command correspond to the number and order of soft keys on IP phones.

You must configure the MeetMe soft key to initiate a meet-me conference. Use this soft key for hardware conferencing only.

Examples

In the following example, ephone template 1 modifies the soft keys in the seized, alerting, and connected call states:

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1

Router(config-ephone-template)# softkeys seized Redial Cfwdall Pickup

Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone.
ephone-template (ephone)	Applies an ephone template to an ephone.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.
softkeys alerting	Modifies the soft keys that display during the alerting call stage.
softkeys connected	Modifies the soft keys that display during the connected call stage.
softkeys idle	Modifies the soft keys that display during the idle call stage.

softkeys seized (voice register template)

To modify the soft key display for the seized call state on Cisco Unified SIP IP phones, use the **softkeys seized** command in voice register template configuration mode. To return to the default, use the **no** form of this command.

softkeys seized [**Cfwdall**] [**Endcall**] [**Gpickup**] [**MeetMe**] [**Pickup**] [**Redial**]
no softkeys seized

Syntax Description

Cfwdall	(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Short for "Call forward all." Forwards all calls.
Endcall	(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Ends the current call.
Gpickup	(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Short for "Group call pick up." Selectively picks up calls coming into a phone number that is a member of a pickup group.
MeetMe	(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Short for "MeetMe conference." Initiates a meet-me conference.
Pickup	(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Short for "Call pick up." Selectively picks up calls coming into another extension.
Redial	(Optional) Appears on the Cisco Unified SIP IP phone during the seized call state. Redials the last number dialed.

Command Default

The default soft keys for the seized call state and the order in which they appear on Cisco Unified SIP IP phones are, from first to last: Cfwdall, Endcall, Gpickup, MeetMe, Pickup, and Redial.

Command Modes

Voice register template configuration (config-register-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
15.2(2)T	Cisco Unified CME 9.0	This command was modified. The **MeetMe** keyword was added.

Usage Guidelines

The seized calling state is when a caller goes offhook before any other action is taken.

The number and order of soft keys used in this command correspond to the number and order of soft keys that appear on Cisco Unified SIP IP phones. Any soft key that is not explicitly specified with this command is disabled.

The MeetMe soft key is added in the seized state when hardware conference is enabled.

This command is not supported on the Cisco Unified 7905, 7912, 7940, and 7960 SIP IP phones.

Examples

In the following example, Cisco Unified SIP IP phone template 1 is configured for the seized and connected call states:

```
Router(config)# voice register template 1
Router(config-register-template)# softkeys seized Redial Cfwdall
Router(config-register-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
softkeys connected (voice register template)	Configures a Cisco Unified SIP IP phone template for soft key display during the connected call state.
softkeys hold (voice register template)	Configures a Cisco Unified SIP IP phone template for soft key display during the hold call state.
softkeys idle (voice register template)	Configures a Cisco Unified SIP IP phone template for soft key display during the idle call state.
template (voice register pool)	Applies a template to a Cisco Unified SIP IP phone.

source-addr

To specify the IP address of the certification authority proxy function (CAPF) server on the Cisco Unified CME router, use the **source-addr** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

source-addr *ip-address*
no source-addr

Syntax Description

ip-address	IP address of the Cisco Unified CME router.

Command Default

No IP address is entered for the CAPF server in the router configuration.

Command Modes

CAPF-server configuration (config-capf-server)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example identifies the IP address for the CAPF server as 10.10.10.1:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

source-address (voice register global)

To identify the IP address and port through which SIP phones communicate with a Cisco CallManager Express (Cisco Unified CME) router, use the **source-address** command in voice register global configuration mode. To disable the router from receiving messages from SIP phones, use the **no** form of this command.

source-address *ip-address* [**port** *port* | **secondary** *ip-address*]
no source-address *ip-address*

Syntax Description

ip-address	Preexisting router IP address, typically one of the addresses of the Ethernet port of the router.
port *port*	(Optional) TCP/IP port number to use for SIP. Range is 2000 to 9999. Default is 5060 for SIP phones.
secondary *ip-address*	Secondary router for Cisco Unified CME. TCP/IP port number is same as the primary Cisco Unified CME router.

Command Default

Port number for SIP: 5060

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
Cisco IOS XE Everest 16.4.1	Cisco Unified CME 11.6	This command was modified to add the keyword: **secondary**.

Usage Guidelines

This command is a mandatory command. The Cisco CallManager Express router cannot communicate with the Cisco CME phones if the IP address is not provided. If the port number is not provided, the SIP default port for is 5060. The IP address is usually the IP address of the Ethernet port to which the phones are connected.

This command enables a router to receive messages from Cisco IP phones through the specified IP address and port.

For systems using ITS V2.1, Cisco CME 3.0, or later versions, the IP phones receive their initial configuration information and phone firmware from the TFTP server associated with the router. The TFTP server address obtained by the Cisco IP phones points to the router IP address. The Cisco IP phones transfer a configuration file called SIPDefault.cnf. This file is automatically generated by the router through the **source-address** and is placed in router memory. The SIPDefault.cnf file contains the IP address that the phones, using the Session Initiation Protocol (SIP), use to register for service. This IP address corresponds to a valid Cisco Unified CME router IP address (and may be the same as the router TFTP server address).

Examples

The following example shows how to set the IP source address and port:

```
Router(config)# voice register global
Router(config-register-global)# source-address 10.6.21.4 port 6000 secondary 10.6.50.6
```

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phones.
file text (voice register global)	Generates ASCII text files for SIP phones.
tftp-path (voice register global)	Specifies the directory to which the provisioning file for SIP phones in a Cisco CallManager Express (Cisco Unified CME) system will be written.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CME or Cisco SIP SRST environment.

speed-dial

To create speed-dial definitions for a Cisco Unified IP phone or analog phone that uses an analog telephone adaptor (ATA) in a Cisco Unified CME system, use the **speed-dial** command in ephone or ephone-template configuration mode. To disable a speed-dial definition, use the **no** form of this command.

speed-dial *speed-tag digit-string* [**label** *label-text*]
no speed-dial *speed-tag*

Syntax Description

speed-tag	Unique sequence number that identifies a speed-dial definition during configuration tasks. Range is from 1 to 33.
digit-string	Digits to be dialed when the speed-dial button is pressed on an IP phone or the digits to be dialed when the associated code is entered from an analog phone with an ATA device. For IP phones, if the first character of this string is the plus sign (+), this speed-dial number is locked and cannot be changed at the phone. If the only character in this string is a pound sign (#), a user-programmable speed-dial button with no speed-dial number attached is defined.
label *label-text*	(Optional) String that contains identifying text to be displayed next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.

Command Default

No speed-dial definitions are created.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)ZJ	Cisco CME 3.0	The number of speed-dial definitions that can be created was increased from 4 to 33. The ability to program speed-dial numbers at the phone and the ability to lock speed-dial numbers were introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)XL	Cisco CME 3.2.1	This command was modified to allow IP phones to access more speed-dial numbers than the number of available buttons on their phones and to allow analog phones to access up to 33 speed-dial numbers.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.

Cisco IOS Release	Cisco Product	Modification
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The *speed-tag* argument in this command is a unique identifier for a speed-dial definition on the phone that is being configured.

This command must be followed by a quick reboot of the phone using the **restart** command.

If you use an ephone template to apply a to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

This command defines speed-dial numbers that are local to the ephone that is being configured. The **directory entry** defines additional, systemwide speed-dial numbers.

IP Phones

For IP phones, speed-dial numbers can be defined by administrators using this command and the *digit-string* argument. The numbers are locked if the *digit-string* argument begins with a plus sign (+). Locked numbers cannot be changed at the phone. Speed-dial definitions without speed-dial numbers (those defined with only a pound sign) and speed-dial instances with unlocked *digit-string* arguments can be changed by users at their IP phones. Changes made to speed-dial definitions are saved in the router nonvolatile random-access memory (NVRAM) configuration after a timer-based delay.

On Cisco Unified IP phones, speed-dial definitions are assigned to available extension buttons that have not been assigned to extensions. Speed-dial definitions are assigned in the order of their identifier (tag) numbers. For example, if you define speed-dial 1, it is assigned to the first phone button that is available after the buttons that have been assigned to extensions. If you have used two buttons for extensions on a phone, speed-dial 1 is assigned to the third physical button on the phone. When you define speed-dial 2, it is assigned to the fourth physical button on the phone, and so on.

If more speed-dial definitions are created than are supported by the IP phone setup, the extra speed-dial configurations can be dialed from IP phones using this procedure:

1. With the phone on-hook, an IP phone user presses a two-digit speed-dial code (that is, 05 for the entry with tag 5). A new soft key, Abbr, appears in the phone display.
2. The phone user picks up the phone handset and presses the Abbr soft key. The full telephone number associated with the speed-dial tag is dialed.

Prior to Cisco IOS Releases 12.3(11)XL and 12.3(14)T, speed-dial entries that were in excess of the number of physical phone buttons available were ignored.

Analog Phones

Analog phone users who use a Cisco ATA-186, Cisco ATA-188, or Cisco VG 224 to connect to a Cisco Unified CME system use a different method to access speed-dial numbers. Analog phone users press the asterisk (*) key and the speed-dial identifier (tag number) to dial a speed-dial number. For instance, an analog phone user presses *1 to speed dial the number that has been programmed as speed-dial 1 on that ephone. Analog phones can have up to 33 local speed-dial numbers programmed by the system administrator. The numbers cannot be programmed from the phone.

Prior to Cisco IOS Releases 12.3(11)XL and 12.3(14)T, analog phones were limited to nine speed-dial numbers.)

Examples

The following example sets speed-dial button 2 to dial the phone user's assistant at extension 5001 and locks the setting so that the phone user cannot change it at the phone:

```
Router(config)# ephone 23
Router(config-ephone)# speed-dial 2 +5001 label "Assistant"
```

Related Commands

	Description
directory entry	Adds a systemwide phone directory entry or speed-dial entry.
restart (ephone)	Performs a fast reboot of a single IP phone in a Cisco Unified CME system.
restart (telephony-service)	Performs a fast reboot of one or all phones in a Cisco Unified CME system.
ephone-template (ephone)	Applies template to ephone configuration.

speed-dial (voice logout-profile and voice user-profile)

To create speed-dial definitions in a user profile or logout profile for Extension Mobility in Cisco Unified CME, use the **speed-dial** command in voice user-profile configuration mode or voice logout-profile configuration mode. To disable a speed-dial definition, use the **no** form of this command.

speed-dial *speed-tag* *number* [**label** *label*] [**blf**]
no speed-dial *speed-tag*

Syntax Description

speed-tag	Unique sequence number that identifies a speed-dial definition during configuration tasks. Range: 1 to 36.
number	Digits to be dialed when the speed-dial button is pressed on an IP phone, or the digits to be dialed when the associated code is entered from an analog phone with an analog telephone adapter (ATA) device. For IP phones, if the first character of this string is the plus sign (+), this speed-dial number is locked and cannot be changed at the phone. If the only character in this string is a pound sign (#), a user-programmable speed-dial button with no speed-dial number attached is defined.
label *label*	(Optional) String that contains identifying text to be displayed next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.
blf	(Optional) Enables Busy Lamp Field (BLF) monitoring for a speed-dial number.

Command Default

No speed-dial definition is created.

Command Modes

Voice logout-profile configuration (config-logout-profile)
Voice user-profile configuration (config-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command in voice user-profile configuration mode to create a speed-dial definition in a user profile for Extension Mobility. A user profile is downloaded to the IP phone when a user is logged into an IP phone that is registered in Cisco Unified CME and enabled for Extension Mobility.

Use this command in voice logout-profile configuration mode to create a speed-dial definition in a logout profile for Extension Mobility. A logout profile is downloaded to the IP phone when no user is logged into an IP phone that is registered in Cisco Unified CME and enabled for Extension Mobility.

For button appearance, Extension Mobility will associate directory numbers and then associates speed-dial definitions in the logout profile or user profile to phone buttons in a sequential manner. If the profile contains more directory and speed-dial numbers than there are buttons on the physical phone to which the profile is downloaded, the remaining numbers in the profile are ignored.

On Cisco Unified IP phones, speed-dial definitions are assigned to available extension buttons that have not been assigned to extensions. Speed-dial definitions are assigned in the order of their identifier (tag) numbers, from 1 to 36.

Examples

The following example shows the configuration for a user profile to be downloaded when the a phone user logs into a Cisco Unified IP phone that is enabled for Extension Mobility. The lines and speed-dial buttons in this profile that are configured on an IP phone after the user logs in depend on the phone type. For example, if the user logs into a Cisco Unified IP Phone 7970, all buttons are configured according to voice-user profile1. However, if the phone user logs into a Cisco Unified IP Phone 7960, all six lines are mapped to phone buttons and the speed dial is ignored because there is no button available for speed dial.

```
pin 12345
user me password pass123
number 2001 type silent-ring
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for Extension Mobility and assigns a logout profile to this phone.
reset (voice logout-profile and voice user-profile)	Performs complete reboot of all IP phones on which a particular logout-profile or user-profile is downloaded.

speed-dial (voice register pool)

To create a speed-dial definition for a Cisco Unified SIP IP phone or analog phone that uses an analog telephone adaptor (ATA) in a Cisco Unified Communications Manager Express (Cisco Unified CME) system, use the **speed-dial** command in voice register pool configuration mode. To disable a speed-dial definition, use the **no** form of this command.

speed-dial *speed-tag* *digit-string* [**label** *label-text*]
no speed-dial *speed-tag*

Syntax Description

speed-tag	Unique sequence number that identifies a speed-dial definition during configuration tasks. Range is 1 to 113.
digit-string	Digits to be dialed when the speed-dial button is pressed on an IP phone or the digits to be dialed when the associated code is entered from an analog phone with an ATA device. For IP phones, if the first character of this string is a plus sign (+), this speed-dial number is locked and cannot be changed at the phone. If the only character in this string is a pound sign (#), a user-programmable speed-dial button with no speed-dial number attached is defined.
label *label-text*	(Optional) Text string that appears next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.

Command Default

No speed-dial definition is created.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
15.2(4)M	Cisco Unified CME 9.1	This command was modified to increase the number of speed-dial configuration to 113.

Usage Guidelines

The **speed-dial** command creates a speed-dial definition for a Cisco Unified SIP IP phone being configured in Cisco Unified CME.

The *speed-tag* argument is a unique identifier for a speed-dial definition on the phone that is being configured. On Cisco Unified IP phones, speed-dial definitions are assigned to available extension buttons that have not been assigned to extensions. Speed-dial definitions are assigned in the order of their identifier numbers.

For example, if you define speed-dial 1, it is assigned to the first phone button that is available after the buttons that are assigned to extensions. If you used two buttons for extensions on a phone, speed-dial 1 is assigned to the third physical button on the phone. When you define speed-dial 2, it is assigned to the fourth physical button on the phone.

For Cisco Unified IP phones, speed-dial numbers can be assigned by the administrator using the *digit-string* argument and can be locked if the *digit-string* argument begins with a plus sign (+). Locked numbers cannot be changed at the phone. Speed-dial instances without speed-dial numbers (those defined with only a pound sign) and speed-dial instances with unlocked *digit-string* arguments can be changed by users at their Cisco Unified IP phones.

If more speed-dial definitions are created than are supported by the IP phone setup, the extra speed-dial configurations are ignored.

Changes made to speed-dial buttons are saved in the router's NVRAM configuration after a timer-based delay.

Analog phone users who use a Cisco ATA-186 or Cisco ATA-188 to connect to Cisco Unified CME systems use a different method to access speed-dial numbers. Instead of pressing a speed-dial button, phone users with ATA devices press the asterisk (*) key and a *speed-tag* number (speed-dial identifier) to dial a speed-dial number. For instance, a phone user with a Cisco ATA-186 presses *1 to dial the number that has been programmed as speed-dial 1 on that phone.

Phones with ATA devices are limited to a maximum of nine speed-dial numbers that must be programmed by the system administrator. The numbers cannot be programmed from the phone. With phones that use ATA devices, system administrators must be sure to tell phone users when speed-dial numbers have been programmed for their phones.

After you configure the **speed-dial** command, restart the phone using the **reset** command.

Examples

The following example shows how to set speed-dial button 2 to dial the head office at extension 5001 and lock the setting so that the phone user cannot change it at the phone:

```
Router(config)# voice register pool 23
Router(config-register-pool)# speed-dial 2 +5001 label "Head Office"
```

The following example shows how to set speed-dial button 13 to dial the sales office extension number (222):

```
Router(config)# voice register pool 3
Router(config-register-pool)# speed-dial 13 222 label "Sales Office"
```

Related Commands

Command	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.
reset (voice register pool)	Performs a complete reboot of a specific SIP phone associated with a Cisco Unified CME system.
voice register pool	Enters voice register pool configuration mode for SIP phones.

srst dn line-mode

To specify line mode for the ephone-dns that are automatically created in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CME router, use the **srst dn line-mode** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst dn line-mode {**dual** | **dual-octo** | **octo** | **single**}
no srst dn line-mode

Syntax Description

dual	SRST fallback ephone-dns are dual-line.
dual-octo	SRST fallback ephone-dns are dual-line or octo-line, depending on the phone type.
octo	SRST fallback ephone-dns are octo-line.
single	SRST fallback ephone-dns are single-line.

Command Default

Default is single-line mode.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(15)XZ	Cisco Unified CME 4.3	The **dual-octo** and **octo** keywords were added.
12.4(20)T	Cisco Unified CME 7.0	This command with the **dual-octo** and **octo** keywords was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies whether ephone-dns that are created during fallback are dual-line, single-line, or octo-line ephone-dns. It applies only to the ephone-dns that are “learned” automatically from ephone configuration information, and not to ephone-dns that are manually configured in Cisco Unified CME.

If you use the **dual-octo** keyword, the type of ephone-dn that Cisco Unified CME in SRST mode creates depends on the phone type. It creates dual-line ephone-dns if the phone type is a Cisco Unified IP Phone 7902 or 7920, or an analog phone connected to the Cisco VG224 or Cisco ATA. It creates octo-line ephone-dns for all other phone types.

Use the **show telephony-service ephone-dn** command to display Cisco Unified CME parameters for ephone-dns.

Examples

The following example specifies dual-line mode for all SRST fallback ephone-dns.

```
telephony-service
 srst dn line-mode dual
```

Related Commands

Command	Description
show telephony-service ephone-dn	Displays parameters for ephone-dns.
srst mode auto-provision	Enables SRST mode for a Cisco Unified CME router.

srst dn template

To specify an ephone-dn template to be used in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CallManager Express (Cisco Unified CME) router, use the **srst dn template** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst dn template *template-tag*
no srst dn template

Syntax Description

template-tag	Identifying number of an existing ephone-dn template. Range is from 1 to 15.

Command Default

No ephone-dn template is specified.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command applies the specified ephone-dn template to all SRST fallback ephone-dns. Ephone-dn templates are created with the **ephone-dn-template** command.

Use the **show telephony-service ephone-dn-template** to display the contents of ephone-dn templates.

Examples

The following example applies ephone-dn template 2 to all SRST fallback ephone-dns.

```
telephony-service
 srst dn template 2
```

Related Commands

Command	Description
ephone-dn-template	Enters ephone-dn-template configuration mode to create an ephone-dn template.
show telephony-service ephone-dn-template	Displays the contents of ephone-dn templates.

srst ephone description

To specify a description to be associated with an ephone in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CallManager Express (Cisco Unified CME) router, use the **srst ephone description** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst ephone description *string*
no srst ephone description

Syntax Description

string	Description to be associated with an ephone. Maximum string length is 100 characters.

Command Default

No description is specified.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **show telephony-service ephone** to display the ephone description to be associated with SRST fallback phones.

Examples

The following example applies a description to all SRST fallback ephones.

```
telephony-service
 srst ephone description srst fallback auto-provision phone
```

The following excerpt displays a time-stamped SRST description for ephone 1:

```
Router# show running-config
ephone 1
 description srst fallback auto-provision phone : Jul 07 2005 17:45:08
 ephone-template 5
 mac-address 100A.7052.2AAE
 button 1:1 2:2
```

Related Commands

Command	Description
show telephony-service ephone	Displays ephone settings.

srst ephone template

To specify an ephone template to be used in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CallManager Express (Cisco Unified CME) router, use the **srst ephone template** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst ephone template *template-tag*
no srst ephone template

Syntax Description

template-tag	Identifying number of an existing ephone template. Range is from 1 to 20.

Command Default

No ephone template is specified.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Ephone templates are created with the **ephone-template** command. This command applies the specified ephone template to all SRST fallback ephones.

Use the **show telephony-service ephone-template** to display the contents of ephone templates.

Examples

The following example applies ephone template 3 to all SRST fallback ephones.

```
telephony-service
 srst ephone template 3
```

Related Commands

Command	Description
ephone-template	Enters ephone-template configuration mode to create an ephone template.
show telephony-service ephone-template	Displays the contents of ephone templates.

srst mode auto-provision

To enable Survivable Remote Site Telephony (SRST) mode for a Cisco Unified CallManager Express (Cisco Unified CME) router, use the **srst mode auto-provision** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst mode auto-provision {**all** | **dn** | **none**}
no srst mode auto-provision

Syntax Description

all	Includes information for learned ephones and ephone-dns in the running configuration.
dn	Includes information for learned ephone-dns in the running configuration.
none	Does not include information for learned ephones or learned ephone-dns in the running configuration. Use this keyword when you want Cisco Unified CME to provide SRST fallback services for Cisco Unified CallManager.

Command Default

SRST mode is disabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command puts a Cisco Unified CME router into SRST mode to provide fallback call-processing services for IP phones that have lost connection to their Cisco Unified CallManager. The phones can be preconfigured manually or the Cisco Unified CME-SRST router can dynamically learn their configuration. The keywords in this command allow you to specify how much of the learned phone configurations you want to include in the running configuration of the Cisco Unified CME-SRST router.

Use the **none** keyword to enable the Cisco Unified CME router to provide SRST fallback services for Cisco Unified CallManager. Use the **dn** or **all** keyword to enable the Cisco Unified CME router to learn the ephone-dn or ephone and ephone-dn configuration from Cisco Unified CallManager and include the information in its running configuration.

Note We do not recommend that you use the **dn** or **all** keyword if you want Cisco Unified CME to provide SRST fallback services. After the Cisco Unified CME-SRST router learns the phone configuration from Cisco Unified CallManager and you save the configuration, the fallback phones are treated as locally configured phones on the Cisco Unified CME-SRST router which can adversely impact the fallback behavior of those phones.

Examples

The following example shows how to enable the Cisco Unified CME router to provide SRST fallback services for phones connected to Cisco Unified CallManager. Information for learned ephone-dns and ephones is not included in the running configuration.

```
telephony-service
 srst mode auto-provision none
```

Related Commands

Command	Description
show telephony-service all	Displays detailed configuration for phones, voice ports, and dial peers in a Cisco Unified CME system.
srst dn line-mode	Specifies line mode for the ephone-dns that are automatically created in SRST mode on a Cisco Unified CME router.

standby username password

To specify that the standby (secondary backup) router XML interface is enabled, use the **standby username password** command in telephony-service configuration mode on the primary router. To disable the XML interface on the secondary backup router, use the **no** form of this command.

standby username *username* **password [0|6]** *password*
no standby username *username* **password [0|6]** *password*

Syntax Description

username	Specifies the username who is authorized to enable the XML interface.
password	Specifies the password to use for access.

Command Default

An authorized user is not named.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption, based on Unified CME password policy.

Usage Guidelines

Use this command to enable the XML interface on the secondary backup router. The username and password must be the same as that used for access to the primary router.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]**. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Examples

The following example enables the XML interface on the secondary backup router:

```
Router(config)# telephony-service
Router(config-telephony)# standby username admin password 1234
```

Related Commands

Command	Description
username password	To assign a login account username and password to a phone user so that the user can log in to the Cisco Unified CME router.

statistics collect

To enable the collection of call statistics for an ephone hunt group, use the **statistics collect** command in ephone-hunt configuration mode. To stop statistics collection and to delete statistics that have been collected, use the **no** form of this command.

statistics collect
no statistics collect

Syntax Description

This command has no arguments or keywords.

Command Default

The default is no call statistics data is collected.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command is used for the collection of call statistics, such as direct calls to hunt group pilot numbers, or calls to the Basic Automatic Call Distribution (B-ACD) and Auto Attendant service. For detailed information, see Cisco Unified CME B-ACD and Tcl Call-Handling Applications .

The **statistics collect** can be used to activate statistics collection for any number of ephone hunt groups.

Statistics collection begins at the time that the **statistics collect** is entered. A maximum of one week (168 hours) of statistics can be stored at a time. You can display the statistics with the **show hunt-group** or transfer statistics automatically to files using TFTP. The **no statistics collect** deletes all statistics that have been collected.

All or some of the statistics can be output with the **show hunt-group** or sent to files automatically using TFTP by using the **hunt-group report url** **hunt-group report every hours** commands.

Examples

The following example enables the collection of call statistics for ephone hunt group 1 and ephone hunt group 2:

```
Router(config)# ephone-hunt 1
Router(config-ephone-hunt)# statistics collect
Router(config)# ephone-hunt 2
Router(config-ephone-hunt)# statistics collect
```

Related Commands

Command	Description
hunt-group report delay hours	Delays the automatic transfer of Cisco CME B-ACD call statistics to a file.

Command	Description
hunt-group report every hours	Sets the hourly interval at which Cisco CME B-ACD call data is automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco CME B-ACD call statistics are to be sent using TFTP.
show ephone-hunt statistics	Displays ephone-hunt configuration information and current status and statistics information.

statistics collect (voice hunt-group)

To enable the collection of call statistics for a voice hunt group, use the **statistics collect** command in voice hunt-group configuration mode. To return to the default, use the **no** form of this command.

statistics collect
no statistics collect

Syntax Description

This command has no arguments or keywords.

Command Default

No configuration statistics can be collected for voice hunt groups.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Examples

The following example shows how to enable the collection of call statistics for voice hunt group 60:

```
Router(config)# voice hunt-group 60
Router(config-voice-hunt-group)# statistics collect
```

Related Commands

Command	Description
statistics collect	Enables the collection of call statistics for an ephone hunt group.

subnet

To define which IP phones are part of an emergency response location (ERL) for the enhanced 911 service, use the **subnet** command in voice emergency response location configuration mode. To remove the subnet definition, use the **no** form of this command.

subnet [{**1** | **2**}] *IPaddress mask*
no subnet [{**1** | **2**}]

Syntax Description

IPaddress	IP address that identifies which IP phones are part of the ERL.
mask	IP subnet mask for the network segment that is part of the ERL.

Command Default

No subnets are defined.

Command Modes

Voice emergency response location configuration (cfg-emrgncy-resp-location)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added to Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to define the groups of IP phones that are part of an ERL. You can create up to 2 different subnets. To include all phones on a single ERL, you can set the subnet mask to 0.0.0.0 to indicate a "catch-all" subnet.

Examples

In the following example, all IP phones with the IP address of 10.X.X.X or 192.168.X.X are automatically associated with this ERL. If one of the phones dials 911, its extension is replaced with 408 555-0100 before it goes to the PSAP. The PSAP will see that the caller's number is 408 555-0100.

```
voice emergency response location 1
 elin 1 4085550100
 subnet 10.0.0.0 255.0.0.0
 subnet 2 192.168.0.0 255.255.0.0
```

Related Commands

Command	Description
elin	Specifies a PSTN number that will replace the caller's extension.

system message

To set a text message for display on idle Cisco IP Phones with display, such as Cisco IP Phone 7940 and Cisco IP Phone 7960, in a Cisco Unified Communications Manager Express (Cisco Unified CME) system, use the **system message** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

system message *text-message*
no system message

Syntax Description

text-message	Alphanumeric string of approximately 30 characters maximum to display when the phone is idle.

Command Default

The message "Cisco Unified Communications Manager Express" is displayed.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco Unified CME 3.0	This command was introduced.
12.3(4)T	Cisco Unified CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The number of characters that can be displayed is not fixed because IP phones typically use a proportional (as opposed to a fixed-width) font. There is room for approximately 30 alphanumeric characters.

The display message is refreshed with a new message after any of the following events occurs:

- A busy phone goes back on-hook.
- An idle phone receives a keepalive message.
- A phone is restarted.

Examples

The following example sets the message "ABC Company" to display instead of "Cisco Unified Communications Manager Express" on idle Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G:

```
Router(config)# telephony-service
Router(config-telephony)# system message ABC Company
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

Cisco Unified CME Commands: T

telephony-service

To enter telephony-service configuration mode for configuring Cisco Unified CME, use the **telephony-service** command in global configuration mode. To remove the entire Cisco Unified CME configuration for SCCP IP phones, use the **no** form of this command.

telephony-service [**setup**]
no telephony-service

Syntax Description

setup	(Optional) Interactive setup tool for configuring Cisco Unified IP Phone 7910s, 7940s, and 7960s in Cisco Unified CME. **Note** This interactive Cisco CME setup tool is restricted to generating basic configuration files for Cisco Unified IP Phone 7910s, 7940s, and 7960s running SCCP protocol only.

Command Default

No Cisco Unified CME configuration for SCCP IP phones is present.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)ZJ	Cisco CME 3.0	The **setup** keyword was added.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command enters the telephony-service configuration mode for configuring system wide parameters for SCCP IP phones in Cisco Unified CME.

Note The voice-gateway system is tied to the telephony service. The **telephony-service** command must be configured before the voice-gateway system is configured; otherwise, the voice gateway is hidden from the user.

Use the **setup** keyword to start the interactive setup tool to automatically configure only Cisco Unified IP Phone 7910s, 7940s, and 7960s in Cisco Unified CME.

For alternate methods of automatically configuring Cisco Unified CME, including Cisco Unified IP Phone 7910s, 7940s, and 7960s and other Cisco Unified IP phones, see the Cisco Unified CME Administrator Guide.

The **setup** keyword is not stored in the router nonvolatile random-access memory (NVRAM).

If you attempt to use the **setup** option for a system that already has a telephony-service configuration, the command is rejected. To use the **setup** option after an existing telephony-service configuration has been created, first remove the existing configuration using the **no telephony-service** command.

The table shows a sample dialog with the Cisco CME setup tool and explains possible responses to the Cisco CME setup tool prompts.

Table 73: Cisco CME Setup Tool Dialog Prompts

Cisco CME Setup Tool Prompt	Description

`Do you want to setup DHCP service for your IP phones? [yes/no]:`
If you respond yes, you see the following prompts:
`IP network for telephony-service DHCP  Pool:`
`Subnet mask for DHCP network :`
`TFTP Server IP address (Option 150) :`
`Default Router for DHCP Pool :`
	• Yes—Configures the Cisco Unified CME router to act as a Dynamic Host Configuration Protocol (DHCP) server, automatically providing IP addresses to your IP phones and provisioning the default gateway and TFTP IP addresses to be used by the phones. This method creates a single pool of IP addresses. If you need a pool for non-IP phones or if the Cisco router cannot act as the DHCP router, answer no and manually define the DHCP server.
	• No—Indicates that you have already configured DHCP or static IP addresses for the IP phones.

`Do you want to start telephony-service setup? [yes/no]:`	• **Yes—**Starts the interactive setup tool for configuring Cisco Unified IP Phone 7910s, 7940s, and 7960s. • **No**—Terminates the Cisco CME setup tool.
`Enter the IP source address for Cisco CallManager Express:` `Enter the Skinny Port for Cisco CallManager Express:  [2000]:`	IP address on which the router provides Cisco Unified CME services, usually the default gateway for the IP subnet that you are using for the IP phones, and the port for Skinny Client Control Protocol (SCCP) messages.
`How many IP phones do you want to configure : [0]:`	Enter the maximum number of IP phones that this Cisco Unified CME system will support. This number can be increased later, to the maximum allowed for this version and your router. **Note** The Cisco CME setup tool associates one number with each newly registering phone. If you want additional numbers on a phone, manually add them later.
`Do you want dual-line extensions assigned to phones? [yes for dual-line / no for single-line]:`	• **Yes**—Each newly registering IP phones is assigned a single number that is associated with a single phone button. The system generates a dual-line ephone-dn entry for each ephone-dn. • **No**—IP phones are linked directly to one or more PSTN trunk lines. Using keyswitch mode requires manual configuration in addition to using the Cisco CME setup tool. The system generates two ephone-dn entries for each ephone-dn, and they are both assigned to a single phone.

Cisco CME Setup Tool Prompt	Description

`What language do you want on IP phones?`
`0 English`
`1 French`
`2 German`
`3 Russian`
`4 Spanish`
`5 Italian`
`6 Dutch`
`7 Norwegian`
`8 Portuguese`
`9 Danish`
`10 Swedish`
`[0]:`
	Language for IP phone displays, selected from the list.
	• Default is 0, English.

`Which Call Progress tone set do you want on IP phones :`
`0 United States`
`1 France`
`2 Germany`
`3 Russia`
`4 Spain`
`5 Italy`
`6 Netherlands`
`7 Norway`
`8 Portugal`
`9 UK`
`10 Denmark`
`11 Switzerland`
`12 Sweden`
`13 Austria`
`14 Canada`
`[0]:`
	Locale for the tone set used to indicate call status or progress, selected from the list.
	• Default is 0, United States.

`What is the first extension number you want to configure :[0]:`	First number in pool of extension numbers to be created for IP phones connected to the Cisco router to be configured. • Starting with this number, remaining extension numbers are automatically configured in a contiguous manner. • This number must be compatible with your telephone number plan and if you use Direct Inward Dialing (DID) service, with public switched telephone network (PSTN) numbering requirements.
`Do you have Direct-Inward-Dial service for all your phones? [yes/no]:`	• Yes—If you have trunk access to public telephone service by ISDN or VoIP for all extension numbers. The system creates an appropriate dial plan. • No—If you have simple analog phone lines only (for example, foreign exchange office [FXO] interfaces) or if you have trunk access for some lines but not all lines.

Cisco CME Setup Tool Prompt	Description
If you answer yes to the previous question, you see the following prompt: `Enter the full E.164 number for the first phone:`	Complete ten-digit telephone number, including area code, that corresponds to the first extension number.
`Do you want to forward calls to a voice message service? [yes/no]:`	• Yes—To forward calls to a single voice message service number when an IP phone is busy or does not answer. All phone extensions forward their calls to the same voice message service pilot number. • No—Not to forward calls to a single voice message service number. Answer no if you do not have a voice message system or if you want to customize call-forwarding behavior for each extension.
If you answer yes to the previous question, you see the following prompt: `Enter the extension or pilot number of the voice message service:`	Voice message service pilot number. • This step can be ignored during the setup dialog and manually configured later.
`Call forward No Answer Timeout: [18]:`	Timeout, in seconds, after which to forward calls to voice mail if they are not answered. • Default is 18.
`Do you wish to change any of the above information? [yes/no]:`	• **Yes**—Starts the dialog over again without implementing any of the answers that you previously gave. • **No**—Uses specified values to automatically build basic configuration for Cisco Unified IP Phone 7910s, 7940s, and 7960s in Cisco Unified CME.

Examples

The following example shows how to enter telephony-service configuration mode for manually configuring Cisco Unified CME. This example also includes the for configuring the maximum number of phones to 12:

```
Router(config)# telephony-service
Router(config-telephony)# max-ephones 12
```

The following example shows how to start the Cisco CME setup tool:

```
Router(config)# telephony-service setup
```

telnet-support

To enable the telnet access for the phone, use the **telnet-support** command in voice register pool-type mode. To disable telnet support, use the **no** form of this command.

telnet-support
notelnet-support

Syntax Description

This command has no arguments or keywords.

Command Default

The telnet support is not enable. When the **reference-pooltype** command is configured, the telnet-support value of the reference phone is inherited.

Command Modes

Voice Register Pool Configuration (config-register-pool)`

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to enable the telnet access for the phone. When you use the no form of this command, the inherited properties of the reference phone takes precedence over the default value.

Example

The following example shows how to specify a description for a phone model using the **description** command:

```
Router(config)# voice register pool-type 9900
Router(config--register-pool-type)# telnet-support
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.

template (auto-register)

To create a basic configuration template that supports all the configurations available on the voice register template, use the **template** command in voice auto register configuration mode. This command is a sub-mode CLI of the command **auto-register** . To disable creation of the basic configuration template as part of the auto registration process , use the **no** form of this command.

template *tag*
no template

Syntax Description

template *tag*	Creates a basic configuration template that supports all the configurations available on the voice register template. Range: 1 to 10.

Command Default

By default, this command is disabled.

Command Modes

voice auto register configuration (config-voice-auto-register)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

This command provides the option to create a basic configuration template that can be applied to all phones registering automatically on Unified CME. It is mandatory that voice register template is configured with the same template tag.

Examples

The following example shows how to create a basic configuration template for auto registration of SIP phones:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)# ?

VOICE auto register configuration commands:
auto-assign Define DN range for auto assignment
default Set a command to its defaults
exit Exit from voice register group configuration mode
no Negate a command or set its defaults
password Default password for auto-register phones
service-enable Enable SIP phone Auto-Registration
template Default template for auto-register phones

Router(config-voice-auto-register)#template ?
 <1-10> template tag
Router(config-voice-auto-register)#template 10
```

Related Commands

Command	Description
service-enable (auto-register)	Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.
password (auto-register)	Configures the mandatory password that administrator sets for auto registration of SIP phones on Unified CME.
auto-assign (auto-register)	Configures the mandatory range of directory numbers for phones auto registering on Unified CME.
auto-register	Enables automatic registration of SIP phones with the Cisco Unified CME system.
auto-reg-ephone	Enables automatic registration of ephones with the Cisco Unified CME system.

template (voice register pool)

To apply a template to a SIP phone, use the **template** command in voice register pool configuration mode. To remove the template, use the **no** form of this command.

template *template-tag*
no template *template-tag*

Syntax Description

template-tag	The template tag that was created with the **voice register template** command in voice register global configuration mode. Range is 1 to 5.

Command Default

Template is not applied to a SIP IP phone.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

Apply any one of five previously defined templates to a SIP phone. Only one template is applied to a SIP phone at one time.

Examples

The following example shows how to define templates 1 and 2 and apply template 1 to SIP phones 1, 2, and 3, and template 2 to SIP phone 4:

```
Router(config)#  voice register template 1
Router(config-register-temp)# anonymous block
Router(config-register-temp)# caller-id block
Router(config-register-temp)# voicemail 5001 timeout 15
Router(config)# voice register template 2

Router(config-register-temp)# anonymous block
Router(config-register-temp)# caller-id block
Router(config-register-temp)# no conference
Router(config-register-temp)# no transfer-attended
Router(config-register-temp)# voicemail 5005 timeout 15
Router(config)# voice register pool 1
Router(config-register-pool)# template 1
Router(config)# voice register pool 2
Router(config-register-pool)# template 1
Router(config)# voice register pool 3
Router(config-register-pool)# template 1
Router(config)# voice register pool 4
Router(config-register-pool)# template 2
```

Related Commands

	Description
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

tftp-path (voice register global)

To specify the directory to which the configuring files for SIP phones in Cisco Unified CME are written, use the **tftp-path** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

tftp-path{**flash:** | **slot0:** | **tftp:**//*url*}
no tftp-path

Syntax Description

flash:	Router flash memory.
slot0:	Router slot 0 memory.
tftp://	External TFTP server.
url	URL for external TFTP server.

Command Default

The default directory is system memory (system:/cme/sipphone/).

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command defines the location for configuration files that are generated by using the **create profile** command.

Examples

The following example shows how to set the path to an HTTP directory for an external TFTP server:

```
Router(config)# voice register global
Router(config-register-global)# tftp-path tftp://mycompany.com/files/
```

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phones.
reset (voice register global)	Performs a “hard” reboot similar to a power-off-power-on sequence for all SIP phones in Cisco Unified CME, including contacting the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information.

tftp-server-credentials trustpoint

To specify the PKI trustpoint that signs the phone configuration files, use the **tftp-server-credentials trustpoint** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

tftp-server-credentials trustpoint *label*
no tftp-server-credentials trustpoint

Syntax Description

label	Name of a configured PKI trustpoint with a valid certificate.

Command Default

No trustpoint is defined for TFTP server communications.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example names the CA trustpoint, server12, as the trustpoint that signs the phone configuration files.

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authenticated
Router(config-telephony)# secure-signaling trustpoint server25
Router(config-telephony)# tftp-server-credentials trustpoint server12
Router(config-telephony)# load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
Router(config-telephony)# exit
```

time-format

To select a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a Cisco CallManager Express (Cisco CME) system, use the **time-format** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

time-format {**12** | **24**}
no time-format

Syntax Description

12	Selects a 12-hour clock. This is the default.
24	Selects a 24-hour clock.

Command Default

Time is displayed in 12-hour clock format.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Examples

The following example selects a 24-hour clock for the time display on Cisco IP phones:

```
Router(config)# telephony-service
Router(config-telephony)# time-format 24
```

Related Commands

	Description
date-format	Selects a format to display the date on Cisco IP phones.

time-format (voice register global)

To set the time display format on SIP phones in a Cisco CallManager Express (Cisco CME) system, use the **timeformat** command in voice register global configuration mode. To display the time in the default format, use the **no** form of this command.

time-format {**12** | **24**}
no **date-format**

Syntax Description

12	Sets time in a 12-hour (AM/PM) clock.
24	Sets time in a 24-hour clock.

Command Default

Time is displayed in 12-hour clock format.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Examples

The following example shows how to set the time format to a 24-hour clock so that 11:00PM is displayed as 2300.

```
Router(config)# voice register global
Router(config-register-global)# time-format 24
```

Related Commands

	Description
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

timeout (ephone-hunt)

To define the number of seconds after which a call that is not answered is redirected to the next number in a hunt-group list in Cisco Unified CME, use the **timeout** command in ephone-hunt configuration mode. To return to the default, use the **no** form of this command.

timeout*seconds[, seconds...]*
no timeout*seconds[,seconds...]*

Syntax Description

seconds	Number of seconds. Range: 3 to 60000. You can enter a different value for each hop between ephone-dns in a hunt group. If you enter a single value, the value is applied to each hop between ephone-dns in a hunt group.

Command Default

Default is the value of the **timeouts ringing** which has a default of 180 seconds if it is not set to another value.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was modified to accept multiple arguments that correspond to the number of ephone-dns configured in the hunt group.
12.4(9)T	Cisco Unified CME 4.0	This command with modifications up was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to set no-answer timeouts for each hop in a hunt group. You can enter a different value for each hop between ephone-dns in a hunt group or you enter a single value to be applied to each hop between ephone-dns in a hunt group list.

If you configure this command and you also configure the **max-timeout** for an ephone hunt group, the **max-timeout** takes precedence over this command.

Examples

The following example defines a no-answer timeout of 10 seconds for each hop between ephone-dns in hunt group 25. If extension 1001 does not answer in 10 seconds, the call is sent to 1002. If 1002 does not answer in 10 seconds, the call is sent to 1003. If 1003 does not answer in 10 seconds, the call is sent to the final number.

```
ephone-hunt 25 sequential
 pilot 4200
 list 1001, 1002, 1003
 timeout 10
 final 4500
```

The following example shows a hunt-group configuration with separate timeouts, one for each ephone in the hunt-group. If the first extension (1001) does not answer in 7 seconds, the call is sent to the second extension (1002). If the call is not answered by the second extension in 9 seconds, the call is forwarded to the third extension (1003). Extension 1003 has 15 seconds to answer before the call is sent to the final number.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 timeout 7, 9, 15
 final 4500
```

The following example shows the configuration for an ephone hunt group for which the **max-timeout** command is also configured. Using this configuration, if the second number is busy, the third extension, 1003, has only 13 seconds to answer (20 - 7 = 13) because the value for max-timeout is 20 seconds.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 timeout 7, 9, 15
 max-timeout 20
 final 4500
```

Related Commands

	Description
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco Unified CME system.
max-timeout	Sets the maximum combined timeout for the no-answer periods for all ephone-dns in an ephone-hunt list,
no-reg (ephone-hunt)	Specifies that the pilot number of an ephone hunt group should not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that callers dial to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.

timeout (voice hunt-group)

To define the number of seconds after which a call that is not answered is redirected to the next number in a voice hunt-group list, use the **timeout** command in voice hunt-group configuration mode. To return to the default timeout, use the **no** form of this command.

timeout *seconds*
no timeout

Syntax Description

seconds	Number of seconds. Range is 3 to 60000. Default is 180.

Command Default

Timeout period is 180 seconds.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

If Call Forward No Answer is configured for a directory number in the voice hunt group, set the timeout value of this command to a value that is less than the timeout value of the **call-forward noan** command.

Examples

The following example shows how to define a no-answer timeout of 15 seconds for each hop between phones in peer hunt-group 25:

```
Router(config)# voice hunt-group 25 peer
Router(config-voice-hunt-group)# timeout 15
```

Related Commands

Command	Description
call-forward noan	Enables call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another number.
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next directory number in a peer voice hunt-group list before proceeding to the final directory number.
list (voice hunt-group)	Defines the directory numbers that participate in a hunt group.

timeouts busy

To set the amount of time after which a call is disconnected from a busy signal, use the **timeouts busy** command in telephony-service configuration mode. To return to the default value, use the **no** form of this command.

timeouts busy *seconds*
no timeouts busy

Syntax Description

seconds	Number of seconds after connection before a call is disconnected from a busy signal. Range is from 0 to 30 seconds. Default is 10.

Command Default

Timeout busy period is 10 seconds.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(8)T	Cisco ITS 2.0	This command was introduced.

Examples

The following example sets a busy timeout of 10 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# timeouts busy 10
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.

timeouts interdigit (telephony-service)

To set the interdigit timeout value for all Cisco IP phones in a Cisco Unified CME system, use the **timeouts interdigit** command in telephony-service configuration mode. To return to the default value, use the **no** form of this command.

timeouts interdigit *seconds*
no timeouts interdigit

Syntax Description

seconds	Interdigit timeout duration for Cisco IP phones, in seconds. Range is from 2 to 120. Default is 10.

Command Default

Timeout period is 10 seconds.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XB	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

The interdigit timeout timer is activated when the caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. This command specifies how long, in seconds, the system waits after a caller enters an initial digit or a subsequent digit of a dialed string. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is terminated. The default is 10 seconds.

To disable the timeouts interdigit timer, set the *seconds* value to zero.

Examples

The following example sets the interdigit timeout value to 5 seconds for all Cisco IP phones:

```
Router(config)# telephony-service
Router(config-telephony)# timeouts interdigit 5
```

In this example, 5 seconds is also the elapsed time after which an incompletely dialed number times out. For example, if you dial nine digits (408555013) instead of the required ten digits (4085550134), you hear a busy tone after 5 “timeout” seconds.

Related Commands

	Description
timeouts interdigit (voice-port)	Configures the interdigit timeout value for a specified voice port.

timeouts interdigit (voice register global)

To set the interdigit timeout value for all Cisco SIP phones in a Cisco Unified CME system, use the **timeouts interdigit** command in voice register global configuration mode. To return to the default value, use the **no** form of this command.

timeouts interdigit *seconds*
no timeouts interdigit

Syntax Description

seconds	Interdigit timeout duration for Cisco SIP phones, in seconds. Range is from 2 to 120. Default is 10.

Command Default

Timeout period is 10 seconds.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1	Cisco Unified CME 11.6	This command was introduced.

Usage Guidelines

The interdigit timeout timer is activated when the caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. This command specifies how long, in seconds, the system waits after a caller enters an initial digit or a subsequent digit of a dialed string. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is terminated. The default is 10 seconds.

To disable the timeouts interdigit timer, set the *seconds* value to zero.

Examples

The following example sets the interdigit timeout value to 5 seconds for all Cisco SIP phones:

```
Router(config)# voice register global
Router(config-register-global)# timeouts interdigit 5
```

Related Commands

	Description
timeouts interdigit (telephoy-service)	Configures the interdigit timeout value for a SCCP phone in Cisco Unified CME system.

timeouts night-service-bell

To specify the interval between two night-service notification bells, use the **timeouts night-service-bell** command in telephony-service configuration mode. To reset to the default value, use the **no** form of this command.

timeouts night-service-bell *seconds*
no timeouts night-service-bell

Syntax Description

seconds	Duration, in seconds, between night-service notification bells. Range: 4 to 30. Default: 12.

Command Default

Default is 12 seconds.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW5	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

This command modifies the repeat interval between two night-service notification bells for the same call from the default (12 seconds) to the specified number of seconds.

When an ephone-dn is marked for night-service treatment, incoming calls that ring during the night-service time period on that directory number send a notification to all IP phones that are marked to receive night-service bell notification.

Examples

The following partial output shows that the night-service notification bell is configured for 4 seconds between bells for the same call:

```
Router# show running-configuration
.
.
.
telephony-service
.
.
.
 night-service code *1234
 night-service day Tue 00:00 23:00
 night-service day Wed 01:00 23:59
 timeouts night-service-bell 4
!
!
```

Related Commands

Command	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received during night-service time periods on ephone-dns that are marked for night service.

Command	Description
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.

timeouts ringing (telephony-service)

To set the timeout value for ringing in a Cisco CallManager Express (Cisco CME) system, use the **timeouts ringing** command in telephony-service configuration mode. To reset the timeout value to the default value, use the **no** form of this command.

timeouts ringing *seconds*
no timeouts ringing

Syntax Description

seconds	Duration, in seconds, for which the Cisco CME system allows ringing to continue if a call is not answered. Range is from 5 to 60000. Default is 180.

Command Default

Timeout is 180 seconds.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following example allows incoming calls to ring for 600 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# timeouts ringing 600
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.

timeouts transfer-recall

To enable Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy, use the **timeouts transfer-recall** command in ephone-dn, ephone-dn template, or telephony-service configuration mode. To reset to the default value, use the **no** form of this command.

timeouts transfer-recall *seconds*
no timeouts transfer-recall

Syntax Description

seconds	Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).

Command Default

Transfer recall is disabled (0 seconds).

Command Modes

Ephone-dn (config-ephone-dn)
Ephone-dn template (config-ephone-dn-template)
Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command enables Call Transfer Recall and sets the number of seconds that Cisco Unified CME waits before sending a transferred call back to the phone that initiated the transfer (transferor).

If the transfer-recall timer set with this command expires before the transfer-to party answers a call, the call is directed back to the transferor and the message, "Transfer Recall From xxxx" displays on the transferor phone. After the first recall, the timer restarts. The maximum number of retries is two if the transfer-to party remains busy or does not answer. The transferor and transfer-to party must be on the same Cisco Unified CME router; the transferee party can be remote.

Transfer recall is not supported if the transfer-to party has Call Forward Busy configured or is a member of any hunt group. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls a transferred call only if the transfer-recall timeout is less than the timeout set with the **call-forward noan** command. If the transfer-recall timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number if the transfer-to party does not answer.

If the transferor is busy at the time of the recall, Cisco Unified CME attempts the recall again after the retry timer expires. The maximum number of retries is two. If the transferor phone remains busy, the call is disconnected after the third recall attempt.

Use this command in telephony-service configuration mode to enable the transfer-recall timer at the system level for all directory numbers. Use this command in ephone-dn configuration mode to enable the transfer-recall timer for a particular directory number, or use the command in ephone-dn template mode to apply it to one or more directory numbers.

If you use an ephone-dn template to apply a command to a directory number and you also use the same command in ephone-dn configuration mode for the same directory number, the value that you set in ephone-dn configuration mode has priority. This command, set in telephony-service configuration mode, has the lowest priority.

Examples

The following example shows that transfer recall is enabled for extension 1030 (ephone-dn 103), which is assigned to ephone 3. If extension 1030 forwards a call and the transfer-to party does not answer, after 60 seconds the unanswered call is sent back to extension 1030 (transferor).

```
ephone-dn  103
 number 1030
 name Smith, John
 timeouts transfer-recall 60
!
ephone  3
 mac-address 002D.264E.54FA
 type 7962
 button  1:103
```

Related Commands

Command	Description
call-forward busy	Enables call forwarding so that incoming calls to a busy extension (ephone-dn) are forwarded to another extension.
call-forward noan	Enables call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another number.
transfer-mode	Specifies the call transfer method for an individual directory number.
transfer-system	Specifies the call transfer method globally for all directory numbers.
trunk	Associates an ephone-dn with a foreign exchange office (FXO) port.

timeouts transfer-recall (voice register global)

To enable Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy, use the **timeouts transfer-recall** command in voice register global configuration mode. To reset to the default value, use the **no** form of this command.

timeouts transfer-recall *seconds*
no timeouts transfer-recall

Syntax Description

seconds	Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).

Command Default

Transfer recall is disabled (0 seconds) on a Cisco Unified SIP IP phone.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1	Cisco Unified CME 11.6	This command was introduced.

Usage Guidelines

This command enables Call Transfer Recall and sets the number of seconds that Cisco Unified CME waits before sending a transferred call back to the phone that initiated the transfer (transferor).

If the transfer-recall timer set with this command expires before the transfer-to party answers a call, the call is directed back to the transferor and the message, “Transfer Recall From xxxx” displays on the transferor phone. If the transferor is busy after the recall, the timer restarts. The maximum number of retries is two if the transfer-to party remains busy or does not answer. The transferor and transfer-to party must be on the same Cisco Unified CME router; the transferee party can be remote.

Transfer recall is not supported if the transfer-to party has Call Forward Busy configured or is a member of any hunt group. The transferor phone and transfer-to phone must be registered to the same Cisco Unified CME, however the transferee phone can be remote. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls the call only if the transfer-recall timeout is set to less than the CFNA timeout. If the transfer-recall timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number after the transfer-to party does not answer. If the transfer-recall timeout is equal to the CFNA timeout, the call is forwarded to the CFNA target number as the CFNA timeout expires before the transfer-recall timeout.

When Call Forward All is configured in Cisco Unified CME, the call is forwarded directly to call forward target number irrespective of whether the phone is busy or idle. In this scenario, transfer recall is not applicable after the call is forwarded.

If the transferor phone is busy, Cisco Unified CME attempts the recall again after the transfer-recall timeout value expires. Cisco Unified CME attempts a recall up to three times. If the transferor phone remains busy, the call is disconnected after the third recall attempt. Also, if the transferor phone is a shared line, and if one of the phones is idle, the transfer recall is directed to the transferor phone that is idle.

Use this command in voice register global configuration mode to enable the transfer-recall timer at the system level for all directory numbers.

The **timeouts transfer-recall** command in voice register global configuration mode has lesser priority than the value that you set in voice register dn configuration mode for the same directory number.

Examples

The following example shows that transfer recall is enabled for 20 seconds. If the transfer-to party does not answer after 20 seconds, the unanswered call is sent back to the (transferor).

```
Router(config)# voice register global
Router(config-register-global)# timeouts transfer-recall 20
```

Related Commands

Command	Description
timeouts transfer-recall(Ephone-dn (config-ephone-dn) and Telephony-service configuration (config-telephony)	Enables Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy.

timeouts transfer-recall (voice register dn)

To enable Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy, use the **timeouts transfer-recall** command in voice register dn configuration mode. To reset to the default value, use the **no** form of this command.

timeouts transfer-recall *seconds*
no timeouts transfer-recall

Syntax Description

seconds	Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).

Command Default

Transfer recall is disabled (0 seconds) on a Cisco Unified SIP IP phone.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1	Cisco Unified CME 11.6	This command was introduced.

Usage Guidelines

This command enables Call Transfer Recall and sets the number of seconds that Cisco Unified CME waits before sending a transferred call back to the phone that initiated the transfer (transferor).

If the transfer-recall timer set with this command expires before the transfer-to party answers a call, the call is directed back to the transferor and the message, “Transfer Recall From xxxx” displays on the transferor phone. If the transferor is busy after the recall, the timer restarts. The maximum number of retries is two if the transfer-to party remains busy or does not answer. The transferor and transfer-to party must be on the same Cisco Unified CME router; the transferee party can be remote.

Transfer recall is not supported if the transfer-to party has Call Forward Busy configured or is a member of any hunt group. The transferor phone and transfer-to phone must be registered to the same Cisco Unified CME, however the transferee phone can be remote. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls the call only if the transfer-recall timeout is set to less than the CFNA timeout. If the transfer-recall timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number after the transfer-to party does not answer. If the transfer-recall timeout is equal to the CFNA timeout, the call is forwarded to the CFNA target number as the CFNA timeout expires before the transfer-recall timeout.

When Call Forward All is configured in Cisco Unified CME, the call is forwarded directly to call forward target number irrespective of whether the phone is busy or idle. In this scenario, transfer recall is not applicable after the call is forwarded.

If the transferor phone is busy, Cisco Unified CME attempts the recall again after the transfer-recall timeout value expires. Cisco Unified CME attempts a recall up to three times. If the transferor phone remains busy, the call is disconnected after the third recall attempt. Also, if the transferor phone is a shared line, and if one of the phones is idle, the transfer recall is directed to the transferor phone that is idle.

Use this command in voice register dn configuration mode to enable the transfer-recall timer for a particular directory number.

If you use the **timeouts transfer-recall** command in voice register dn configuration mode for the same directory number, the value that you set in voice register dn configuration mode has priority than the value set in the voice register global configuration mode (this has the lowest priority).

Examples

The following example shows that transfer recall is enabled for extension 111 (voice register dn 1). If extension 111 forwards a call to voice register dn 2 and the transfer-to party does not answer, after 20 seconds the unanswered call is sent back to extension 1111 (transferor).

```
voice register dn  1
 timeouts transfer-recall 20
 number 111
voice register dn  2
 number 222
```

Related Commands

Command	Description
timeouts transfer-recall(Ephone-dn (config-ephone-dn) and Telephony-service configuration (config-telephony)	Enables Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy.

time-webedit (telephony-service)

To enable the system administrator to set time on the Cisco Unified CME router through the web interface, use the **time-webedit** command in telephony-service configuration mode. To disable this feature, use the **no** form of this command.

time-webedit
no time-webedit

Syntax Description

This command has no arguments or keywords.

Command Default

Time-setting through the web interface is disabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Usage Guidelines

The **time-webedit** allows a local administrator of the Cisco Unified CME router to change and set time through the web-based graphical user interface (GUI).

Note Cisco discourages this method for setting network time. The router should be set up to automatically synchronize its router clock from a network-based clock source using Network Time Protocol (NTP). In the rare case that a network NTP clock source is not available, the **time-webedit** can be used to allow manual setting and resetting of the router clock through the Cisco CME GUI.

Examples

The following example enables web editing of time:

```
Router(config)# telephony-service
Router(config-telephony)# time-webedit
```

Related Commands

	Description
dn-webedit	Enables adding of directory numbers through a web interface.
telephony-service	Enters telephony-service configuration mode.

time-zone

To set the time zone so that the correct local time is displayed on SCCP Cisco Unified IP phones, use the **time-zone** command in telephony-service configuration mode. To disable a time-zone setting configured with the **time-zone** command and return to the default time zone (Pacific Standard Time), use the **no** form of this command.

time-zone *number*
no **time-zone**

Syntax Description

number	Numeric code for a named time zone. The following are the selections. The numbers in parentheses indicate the offset from Coordinated Universal Time (UTC) in minutes. **Note** The time shows incorrectly for phones configured in West Africa during the Summer Time. For West Africa, Summer Time or Daylight Savings Time (DST) is not used. There is no correct time zone in this time zone list to account for this time zone. • **1**—Dateline Standard Time (-720) • **2**—Samoa Standard Time (-660) • **3**—Hawaiian Standard Time (-600) • **4**—Alaskan Standard/Daylight Time (-540) • **5**—Pacific Standard/Daylight Time (-480) • **6**—Mountain Standard/Daylight Time (-420) • **7**—United States (US) Mountain Standard Time (-420) • **8**—Central Standard/Daylight Time (-360) • **9**—Mexico Standard/Daylight Time (-360) • **10**—Canada Central Standard Time (-360) • **11**—SA Pacific Standard Time (-300) • **12**—Eastern Standard/Daylight Time (-300) • **13**—US Eastern Standard Time (-300) • **14**—Atlantic Standard/Daylight Time (-240) • **15**—South America (SA) Western Standard Time (-240) • **16**—Newfoundland Standard/Daylight Time (-210) • **17**—SA Standard/Daylight Time (-180) • **18**—SA Eastern Standard Time (-180) • **19**—Mid-Atlantic Standard/Daylight Time (-120) • **20**—Azores Standard/Daylight Time (-60) • **21**—UTC Standard/Daylight Time (+0) • **22**—Greenwich Standard Time (+0) • **23**—Western Europe Standard/Daylight Time (+60) • **24**—GTB (Athens, Istanbul, Minsk) Standard/Daylight Time (+60) • **25**—Egypt Standard/Daylight Time (+60) • **26**—Eastern Europe Standard/Daylight Time (+60)

number continued	• **27**—Romance Standard/Daylight Time (+120) • **28**—Central Europe Standard/Daylight Time (+120) • **29**—South Africa Standard Time (+120) • **30**—Jerusalem Standard/Daylight Time (+120) • **31**—Saudi Arabia Standard Time (+180) • **32**—Russian Standard/Daylight Time (+180) • **33**—Iran Standard/Daylight Time (+210) • **34**—Caucasus Standard/Daylight Time (+240) • **35**—Arabian Standard Time (+240) • **36**—Afghanistan Standard Time (+270) • **37**—West Asia Standard Time (+300) • **38**—Ekaterinburg Standard Time (+300) • **39**—India Standard Time (+330) • **40**—Central Asia Standard Time (+360) • **41**—Southeast Asia Standard Time (+420) • **42**—China Standard/Daylight Time (+480) • **43**—Taipei Standard Time (+480) • **44**—Tokyo Standard Time (+540) • **45**—Central Australia Standard/Daylight Time (+570) • **46**—Australia Central Standard Time (+570) • **47**—East Australia Standard Time (+600) • **48**—Australia Eastern Standard/Daylight Time (+600) • **49**—West Pacific Standard Time (+600) • **50**—Tasmania Standard/Daylight Time (+600) • **51**—Central Pacific Standard Time (+660) • **52**—Fiji Standard Time (+720) • **53**—New Zealand Standard/Daylight Time (+720) • 54—Venezuela Standard Time (-270) • 55—Pacific SA Daylight Time (-180) • 56—Pacific SA Standard Time (-240)

Command Default

The default is time-zone 5, Pacific Standard/Daylight Time (-480).

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command works with the vendorConfig section of the Sep*.cnf.xml configuration file, which is read by the phone firmware when the Cisco IP Phone is booted up. Certain phones, such as the Cisco Unified IP Phone

7906 , 7911, 7931, 7941, 7942, 7945, 7961, 7962, 7965, 7970 , 7971, and 7975, obtain Coordinated Universal Time (UTC) from the clock of the Cisco router. To display the correct local time, the time display on these phones must be offset by using this command.

This command is not required for Cisco Unified IP Phone 7902G, 7905G, 7912G, 7920, 7921, 7935, 7936, 7940, 7960, or 7985G.

For changes to the time-zone settings take effect, the Sep*.cnf.xml file must be updated by using the **create cnf-files** command and the Cisco IP phones must rebooted by using the **reset** command.

Examples

The following example sets the Cisco IP Phone 7970 units to Fiji Standard Time:

```
Router(config)# telephony-service
Router(config-telephony)# time-zone 53
```

Related Commands

Command	Description
create cnf-files	Sets display and phone functionality for the Cisco IP Phone 7970 units using the vendorConfig parameters of the downloaded firmware’s Sep*.cnf.xml configuration file.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.

timezone (voice register global)

To set the time zone used for SIP phones in a Cisco Unified CME system, use the **timezone** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

timezone *number*
no timezone

Syntax Description

number	Range is 1 to 53. Default is 5, Pacific Standard/Daylight Time

Command Default

Default is 5, Pacific Standard/Daylight Time.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

The following table lists the supported time zone numbers and the corresponding description.

Table 74: Time Zones

Number	Description	Offset in Minutes
1	Dateline Standard Time	-720
2	Samoa Standard Time	-660
3	Hawaiian Standard Time	-600
4	Alaskan Standard/Daylight Time	-540
5	Pacific Standard/Daylight Time	-480
6	Mountain Standard/Daylight Time	-420
7	US Mountain Standard Time	-420
8	Central Standard/Daylight Time	-360
9	Mexico Standard/Daylight Time	-360
10	Canada Central Standard Time	-360
11	SA Pacific Standard Time	-300
12	Eastern Standard/Daylight Time	-300
13	US Eastern Standard Time	-300
14	Atlantic Standard/Daylight Time	-240

Number	Description	Offset in Minutes
15	SA Western Standard Time	-240
16	Newfoundland Standard/Daylight Time	-210
17	South America Standard/Daylight Time	-180
18	SA Eastern Standard Time	-180
19	Mid-Atlantic Standard/Daylight Time	-120
20	Azores Standard/Daylight Time	-60
21	GMT Standard/Daylight Time	+0
22	Greenwich Standard Time	+0
23	W. Europe Standard/Daylight Time	+60
24	GTB Standard/Daylight Time	+60
25	Egypt Standard/Daylight Time	+60
26	E. Europe Standard/Daylight Time	+60
27	Romance Standard/Daylight Time	+120
28	Central Europe Standard/Daylight Time	+120
29	South Africa Standard Time	+120
30	Jerusalem Standard/Daylight Time	+120
31	Saudi Arabia Standard Time	+180
32	Russian Standard/Daylight Time	+180
33	Iran Standard/Daylight Time	+210
34	Caucasus Standard/Daylight Time	+240
35	Arabian Standard Time	+240
36	Afghanistan Standard Time	+270
37	West Asia Standard Time	+300
38	Ekaterinburg Standard Time	+300
39	India Standard Time	+330
40	Central Asia Standard Time	+360
41	SE Asia Standard Time	+420
42	China Standard/Daylight Time	+480

Number	Description	Offset in Minutes
43	Taipei Standard Time	+480
44	Tokyo Standard Time	+540
45	Cen. Australia Standard/Daylight Time	+570
46	AUS Central Standard Time	+570
47	E. Australia Standard Time	+600
48	AUS Eastern Standard/Daylight Time	+600
49	West Pacific Standard Time	+600
50	Tasmania Standard/Daylight Time	+600
51	Central Pacific Standard Time	+660
52	Fiji Standard Time	+720
53	New Zealand Standard/Daylight Time	+720
54	Venezuela Standard Time	-270
55	Pacific SA Daylight Time	-180
56	Pacific SA Standard Time	-240

Examples

The following example shows how top set the time zone to 8, Central Standard Daylight Time:

```
Router(config)# voice register global
Router(config-register-global)# timezone 8
```

Related Commands

Command	Description
dst (voice register global)	Sets the time period for daylight saving time on SIP phones.
dst auto-adjust (voice register global)	Enables automatic adjustment of daylight saving time on SIP phones.
time-format (voice register global)	Selects a 12-hour clock or a 24-hour clock for the time display format on SIP phones in a Cisco CME system

transfer max-length

To specify the maximum length of the transfer number, use the **transfer max-length** command in voice register pool or voice register template configuration mode. To disable the maximum length, use the **no** form of this command.

transfer max-length *max-length*
no transfer max-length *max-length*

Syntax Description

max-length	Maximum length of the transfer number. Range is 3 to 16.

Command Default

No maximum length is specified for the transfer number.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration ((config-register-temp)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

The **transfer max-length** command is used to indicate the maximum length of the number being dialed for a call transfer. When only a specific number of digits are to be allowed during a call transfer, a value between 3 and 16 is configured.When the number dialed exceeds the maximum length configured, then the call transfer is blocked.

Examples

The following example shows how to configure the maximum length of the transfer number under voice register pool 1. Because the maximum length is configured as 5, only call transfers to Cisco Unified SIP IP phones with a five-digit directory number are allowed. All call transfers to directory numbers with more than five digits are blocked.

```
Router# configure terminal
Router(config)# voice register pool 1
Router(config-register-pool)# transfer max-length 5
```

The following example shows how to configure the maximum length of the transfer number for a set of phones under voice register template 2:

```
Router# configure terminal
Router(config)# voice register template 2
Router(config-register-temp)# transfer max-length 10
```

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for a SIP IP phone in Cisco Unified CME or for a set of SIP phones in Cisco Unified SIP SRST
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

transfer-attended (voice register template)

To enable a soft key for attended transfer in a SIP phone template, use the **transfer-attended** command in voice register template configuration mode. To disable the soft key, use the **no** form of this command.

transfer-attended
no transfer-attended

Syntax Description

This command has no arguments or keywords.

Command Default

Soft key is enabled.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command enables a soft key for attended transfer in the specified template which can then be applied to SIP phones in Cisco CME. The attended transfer soft key is enabled by default. To disable the attended transfer soft key, use the **no transfer-attended** command. To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

A blind transfer is one in which the transferring extension connects the caller to a destination extension before ringback begins. A attended transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

Examples

The following example shows how to disable attended transfer in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# no transfer-attended
```

Related Commands

	Description
conference (voice register template)	Enables the soft key for conference in a SIP phone template.
template	Applies a template to a SIP phone.
transfer-blind (voice register template)	Enables a soft key for blind transfer in a SIP phone template.

transfer-blind (voice register template)

To enable a soft key for blind transfer in a SIP phone template, use the **transfer-blind** command in voice register template configuration mode. To disable the soft key, use the **no** form of this command.

transfer-blind
no transfer-blind

Syntax Description

This command has no arguments or keywords.

Command Default

Soft key is enabled.

Command Modes

Voice register template configuration (config-register-template)

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command enables a soft key for blind transfer in the specified template which can then be applied to SIP phones in Cisco CME. The blind transfer soft key is enabled by default. To disable the blind transfer soft key, use the **no transfer-blind** command. To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

A blind transfer is one in which the transferring extension connects the caller to a destination extension before ringback begins. A attended transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

Examples

The following example shows how to disable blind transfer in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# no transfer-blind
```

Related Commands

	Description
conference (voice register template)	Enables the soft key for conference in a SIP phone template.
template	Applies a template to a SIP phone.
transfer-attended (voice register template)	Enables the soft key for attended transfer on SIP phones.

transfer-digit-collect

To select the digit-collection method for consultative call-transfers, use the **transfer-digit-collect** command in telephony-service configuration mode for Cisco Unified CME or in call-manager-fallback configuration mode for Cisco Unified SRST. To reset to the default value, use the **no** form of this command.

transfer-digit-collect {**new-call** | **orig-call**}
no transfer-digit-collect

Syntax Description

new-call	Dialed digits are collected from new call leg. Default value.
orig-call	Dialed digits are collected from original call leg.

Command Default

Digits are collected from the new consultative call-leg (**new-call** keyword).

Command Modes

Telephony-service configuration (config-telephony)

Call-manager-fallback configuration (config-cm-fallback)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies whether the dialed digits of the target number are collected on the original call leg or on the new call leg that is created when a phone user initiates a consultative call-transfer.

For consultative transfers, a local number is matched on the **number** command in ephone-dn configuration mode; a PSTN number is matched on the **transfer-pattern** command in telephony service mode.

The **orig-call** keyword selects the method used in versions before Cisco Unified CME 4.3 and Cisco Unified SRST 4.3. After a phone user presses the Transfer soft key, the dialed digits of the target number are collected on the original call leg and buffered until either a local ephone-dn or transfer-pattern is matched. When the transfer-to number is matched, the original call is put on hold and an idle line or channel is seized to send the dialed digits from the buffer.

The **new-call** keyword selects the default method that is used in Cisco Unified CME 4.3 and later versions and Cisco Unified SRST 4.3 and later versions. The transfer-to digits are collected on a new consultative call-leg that is created when the user presses the Transfer soft key. The consultative call-leg is seized and the dialed digits are passed on without being buffered until the digits are matched and the consultative call-leg moves to the alerting state.

The **new-call** keyword is supported only if:

- The **transfer-system full-consult** command (default) is set in telephony-service configuration mode.
- The **transfer-mode consult** command (default) is set for transferor's directory number (ephone-dn).
- An idle line or channel is available for seizing, digit collection, and dialing.

A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

Examples

The following example shows the digit-collection set to the method used in versions before Cisco Unified CME 4.3 and Cisco Unified SRST 4.3:

```
Router(config)# telephony-service
Router(config-telephony)# transfer-digit-collect orig-call
```

Related Commands

Command	Description
transfer-mode	Specifies the type of call transfer for an individual directory number that uses the ITU-T H.450.2 standard.
transfer-pattern (telephony-service)	Allows the transfer of calls to phones outside the local Cisco Unified CME network.
transfer-system	Specifies the call transfer method for all IP phones on a Cisco Unified CME router using the ITU-T H.450.2 standard.

transfer-mode

To specify the type of call transfer for an individual IP phone extension that uses the ITU-T H.450.2 standard, use the **transfer-mode** command in ephone-dn configuration mode. To remove this specification, use the **no** form of this command.

transfer-mode {**blind** | **consult**}
no transfer-mode

Syntax Description

blind	Transfers calls without consultation using a single phone line.
consult	Transfers calls with consultation using a second phone line, if available.

Command Default

The ephone-dn uses the transfer-system value that was set systemwide.

Command Modes

Ephone-dn configuration (config-ephone)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command specifies the type of call transfer for an individual Cisco IP phone extension that is using the ITU-T H.450.2 protocol. It allows you to override the system default **transfer-system** setting (full-consult or full-blind) for that extension.

Call transfers that use H.450.2 can be blind or consultative. A blind transfer is one in which the transferring phone connects the caller to a destination extension before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

You can specify blind or consultative transfer on a system-wide basis by using the **transfer-system** command. The system-wide setting can then be overridden for individual phone extensions by using the **transfer-mode** command. For example, in a Cisco CallManager Express (Cisco CME) network that is set up for consultative transfer, a specific extension with an auto-attendant that automatically transfers incoming calls to specific extension numbers can be set to use blind transfer, because auto-attendants do not use consultative transfer.

Use this command with Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version.

Examples

The following example sets blind mode for call transfers from this directory number:

```
Router(config)# ephone-dn 21354
Router(config-ephone-dn)# transfer-mode blind
```

Related Commands

	Description
ephone-dn	Enters ephone-dn configuration mode to set directory numbers and parameters for individual Cisco IP phone lines.
transfer-system	Specifies the call transfer method for all IP phones on a Cisco ITS router using the ITU-T H.450.2 standard.

transfer-park blocked

To prevent extensions on an ephone from parking calls, use the **transfer-park blocked** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

transfer-park blocked
no transfer-park blocked

Syntax Description

This command has no arguments or keywords.

Command Default

Transfer to park is allowed.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command prevents transfers to park that use the Trnsfer soft key and a call-park slot number, while allowing call-parks that use only the Park soft key. To prevent use of the Park soft key, use an ephone template to remove it from the phone.

An exception to this is made for phones with dedicated park slots. If the **transfer-park blocked** command is used on an ephone that has a dedicated park slot, the phone is blocked from parking calls at park slots other than the dedicated park slot, but is still able to park calls at its own dedicated park slot. On an IP phone, the user presses the Trnsfer soft key and the call-park feature access code (FAC) to park a call at the phone's dedicated park slot. On an analog phone, the user presses hookflash and the call-park FAC.

When the **transfer-park blocked** command is used on an ephone that does not have a dedicated park slot, the phone is blocked from parking any calls.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example prevents ephone 25 and extensions 234, 235, and 236 from parking calls at any call-park slot.

```
ephone-dn 11
 number 234
ephone-dn 12
 number 235
ephone-dn 13
 number 236
ephone 25
 button 1:11 2:12 3:13
 transfer-park blocked
The following example uses an ephone template to prevent ephone 26 and extension 76589 from
 parking calls at any call-park slot.
ephone-dn 33
```

```
 number 76589
ephone-template 1
 transfer-park blocked
ephone 26
 button 1:33
 ephone-template 1
```

The following example sets up a dedicated park slot for the extensions on ephone 6 and blocks transfers to call park from extensions 2977, 2978, and 2979 on that phone. Those extensions can still park calls at the phone's dedicated park slot by using the Park soft key or Trnsfer and the call-park FAC.

```
ephone-dn 3
 number 2558
 name Park 2977
 park-slot reserved-for 2977 timeout 60 limit 3 recall alternate 3754
ephone-dn 4
 number 2977
ephone-dn 5
 number 2978
ephone-dn 6
 number 2979
ephone 6
 button 1:4 2:5 3:6
 transfer-park blocked
```

transfer-pattern (telephony-service)

To allow transfer of telephone calls from Cisco IP phones to phones other than Cisco IP phones, use the **transfer-pattern** command in telephony-service configuration mode. To disable these transfers, use the **no** form of this command.

transfer-pattern *transfer-pattern* [**blind**]
no transfer-pattern

Syntax Description

transfer-pattern	String of digits for permitted call transfers. Wildcards are allowed. A maximum of 32 transfer patterns can be entered, using a separate command for each one.
blind	(Optional) When H.450.2 consultative call transfer is used, this keyword forces transfers that match the pattern to be executed as blind transfers. Overrides settings made using the **transfer-system** and **transfer-mode** commands.

Command Default

Transfer of calls is enabled only to local Cisco IP phones.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)T	Cisco ITS 2.1	The **blind** keyword was added.

Usage Guidelines

This command allows you to transfer calls to “other” phones—that is, to non-IP phones and phones outside of your network. A call is then established between the transferred party and the new recipient. By default, all Cisco IP phone extension numbers are allowed as transfer targets.

The **blind** keyword is valid only for systems that use Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version and applies only to consultative transfers made using the H.450.2 standard. The **blind** keyword forces calls that are transferred to numbers that match the transfer pattern to be executed as blind or full-blind transfers, overriding any settings made using the **transfer-system** and **transfer-mode** commands.

When defining transfers to non-local numbers, it is important to note that transfer-pattern digit matching is performed before translation-rule operations. Therefore, you should specify in this command the digits actually entered by phone users before they are translated.

Use of the .T control character for the *transfer-pattern* argument is not recommended. The .T control character indicates a variable-length dial string, which causes Cisco CME to wait for an interdigit timeout (default is 10 seconds) before transferring a call. To avoid the interdigit timeout, a matching transfer pattern should be used with the **transfer-pattern** command. For example, use the **transfer-pattern 9.........** command instead of the **transfer-pattern .T** command.

Examples

The following example sets a transfer pattern. A maximum of 32 transfer patterns can be entered. In this example, 55501.. (the two periods are wildcards) permits transfers to any number in the range from 555-0100 to 555-0199.

```
Router(config)# telephony-service
Router(config-telephony)# transfer-pattern 55501..
```

Related Commands

	Description
transfer-mode	Specifies the type of call transfer for an individual IP phone extension number that uses the ITU-T H.450.2 standard.
transfer-system	Specifies the call transfer method for all Cisco CME extensions that use the ITU-T H.450.2 standard.

transfer-pattern blocked

To block all call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone, use the **transfer-pattern blocked** command in voice register pool and voice register template configuration mode. To allow call transfers, use the **no** form of this command.

transfer-pattern blocked
no transfer-pattern blocked

Syntax Description

This command has no arguments or keywords.

Command Default

Call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone are allowed.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration ((config-register-temp)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

When the **transfer-pattern blocked** command is configured for a specific phone, no call transfers are allowed from that phone over the trunk.

This feature forces unconditional blocking of all call transfers from a specific phone to any other non-local numbers (external calls from one trunk to another trunk). No call transfers from this specific phone are possible even when a transfer pattern matches the dialed digits for transfer.

Examples

The following example shows how to block all call transfers for voice register pool 5:

```
Router(config)# voice register pool 5
Router(config-register-pool)# transfer-pattern ?
  blocked  global transfer pattern not allowed
Router(config-register-pool)# transfer-pattern blocked
```

The following example shows how to block all call transfers for a set of Cisco Unified SIP IP phones defined by voice register template 9:

```
Router(config)# voice register template 9
Router(config-register-temp)# transfer-pattern ?
  blocked  global transfer pattern not allowed
Router(config-register-temp)# transfer-pattern blocked
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for a Cisco Unified SIP IP phone in Cisco Unified CME or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST.

Command	Description
voice register template	Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.

transfer-system

To specify the call transfer method to be used by Cisco Unified IP phones in Cisco Unified CME, use the **transfer-system** command in telephony-service configuration mode. To disable the call transfer method, use the **no** form of this command.

transfer-system {**blind** | **full-blind** | **full-consult** [**dss**] | **local-consult**}
no transfer-system

Syntax Description

blind	Transfers calls without consultation using a single phone line and the Cisco proprietary method. This is the default for Cisco CME 3.4 and earlier versions.
full-blind	Transfers calls without consultation using H.450.2 standard methods.
full-consult	Transfers calls using H.450.2 with consultation using a second phone line, if available. The calls fall back to **full-blind** if a second line is not available. This is the default for Cisco Unified CME 4.0 and later versions.
dss	Transfers calls with consultation to idle monitor lines. All other call-transfer behavior is identical to full-consult.
local-consult	Transfers calls with local consultation using a second phone line, if available, or the calls fall back to blind if the target for consultation or transfer is not local. This mode is intended for use primarily in Voice over Frame Relay (VoFR) networks, because the Cisco VoFR call transfer protocol does not support an end-to-end transfer-with-consultation mechanism. Not supported if transfer-to destination is on the Cisco ATA, Cisco VG224, or a SCCP-controlled FXS port.

Command Default

For Cisco Unified CME 4.0 and later versions, the transfer mode is **full-consult**. For Cisco CME 3.4 and earlier versions, the transfer mode is **blind**.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.3(11)T	Cisco CME 3.2	The **dss** keyword was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The command default was changed from **blind** to **full-consult**.
12.4(9)T	Cisco Unified CME 4.0	This command with the default of **full-consult** was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Direct station select is a functionality that allows a multibutton phone user to transfer calls to an idle monitor line by pressing the Transfer key and the appropriate monitor button. The **dss** keyword permits consultative call transfer to monitored lines.

Call transfers can be blind or consultative. A blind transfer is one in which the transferring extension connects the caller to a destination extension before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

The **transfer-system** command specifies whether the H.450.2 standard or a Cisco proprietary method will be used to communicate call transfer information across the network. When you specify use of the H.450.2 consultative or blind mode of transfer globally by using the **transfer-system** command (or by using the default), you can override this mode for individual ephones by using the **transfer-mode** command. For example, in a system that is set up for consultative transfer, a specific extension with an auto-attendant that automatically transfers incoming calls to specific extension numbers can be set to use blind transfer, because auto-attendants do not use consultative transfer.

Prior to Cisco Unified CME 4.0, the default for this command specified the Cisco proprietary method. In Cisco Unified CME 4.0, the default was changed to specify the H.450.2 standard as the transfer method. Check the following table for configuration recommendations for different versions of Cisco Unified CME.

Table 75: Transfer Method Recommendations

Cisco Product	transfer-system Default	transfer-system to Use	Transfer Method Recommendation
Cisco Unified CME 4.0 and later versions	**full-consult**	**full-consult** or**full-blind**	Use H.450.2 for call transfer. Because this is the default for this version, you do not need to use the **transfer-system** command unless you want to use the **full-blind** or **dss** keyword. Optionally, you can use the proprietary Cisco method by using the **transfer-system** command with the **blind** or **local-consult** keyword.
Cisco CME 3.0 to 3.3	**blind**	**full-consult** or**full-blind**	Use H.450.2 for call transfer. You must explicitly configure the **transfer-system** command with the **full-consult** or **full-blind** keyword because H.450.2 is not the default for this version. Optionally, you can use the proprietary Cisco method by using the **transfer-system** command with the **blind** or **local-consult** keyword.
Cisco ITS 2.1 to 3.0	**blind**	**blind** or**local-consult**	Use the Cisco proprietary method. Because this is the default for this version, you do not need to use the **transfer-system** command unless you want to use the **local-consult** keyword. Optionally, you can use the H.450.2 standard for call transfer by using **transfer-system** command with the **full-consult** or **full-blind** keyword. You must also configure the router with a Tcl script that is contained in the file called app-h450-transfer.x.x.x.x.zip. This file is posted on the Cisco Unified CME software download website at http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp .

Examples

The following example sets full consultation as the call transfer method:

```
Router(config)# telephony-service
Router(config-telephony)# transfer-system full-consult
```

Related Commands

	Description
transfer-mode	Specifies the type of call transfer for an individual IP phone extension that uses the H.450.2 standard.

translate (ephone-dn)

To apply a translation rule in order to manipulate the digits that are dialed by users of Cisco Unified IP phones, use the **translate** command in ephone-dn or ephone-dn-template configuration mode. To disable the translation rule, use the **no** form of this command.

translate {**called** | **calling**} *translation-rule-tag*
no translate {**called** | **calling**}

Syntax Description

called	Translate the called number.
calling	Translate the calling number.
translation-rule-tag	Unique sequence number by which the rule set is referenced. This number is arbitrarily chosen. Range is from 1 to 2147483647. There is no default value.

Command Default

No translation rule is applied.

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-template-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command allows you to select a preconfigured translation rule to modify the number dialed by a specific extension (Cisco Unified IP phone destination number, or ephone-dn). A translation rule is a general-purpose digit-manipulation mechanism that performs operations such as automatically adding telephone area and prefix codes to dialed numbers. The translation rules are applied to the voice ports created by the ephone-dn. The **called** keyword translates the called number, and the **calling** keyword translates the calling number.

The translation rule mechanism inserts a delay into the dialing process when digits are entered that do not explicitly match any of the defined translation rules. This delay is set by the interdigit timeout. The translation-rule mechanism uses the delay to ensure that it has acquired all of the digits from the phone user before making a final decision that there is no translation-rule match available (and therefore no translation operation to perform). To avoid this delay, it is recommended that you include a dummy translation rule to act as a pass-through rule for digit strings that do not require translation. For example, a rule like “^5 5” that maps a leading 5 digit into a 5 would be used to prevent the translation rule delay being applied to local extension numbers that started with a 5.

Note For this command to take effect, appropriate translation rules must have been created at the VoIP configuration level. Use the **show voice translation-rule** command to view the translation rules that you have defined. For in formation, see the Dial Peer Configuration on Voice Gateway Routers.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example applies translation rule 20 to numbers called by extension 46839:

```
Router(config)# translation-rule 20
Router(config-translate)# rule 0 1234 2345 abbreviated
Router(config-translate)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 46839
Router(config-ephone-dn)# translate called 20
```

The following example uses an ephone-dn-template to apply translation rule 20 to numbers called by extension 46839:

```
Router(config)# translation-rule 20
Router(config-translate)# rule 0 1234 2345 abbreviated
Router(config-translate)# exit
Router(config)# ephone-dn-template 1
Router(config-ephone-dn-template)# translate called 20
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 46839
Router(config-ephone-dn)# ephone-dn-template 1
```

Related Commands

	Description
rule	Defines a translation rule.
translation-rule	Creates a translation identifier and enters translation-rule configuration mode.

translate callback-number

To assign a translation profile for incoming or outgoing call legs on a Cisco IP phone, use the **translation-profile** command in call-manager-fallback configuration mode. To delete the translation profile from the voice port, use the **no** form of this command.

translate callback-number
no translate callblack-number

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
outgoing	Specifies that this translation profile handles outgoing calls.
name	Name of the translation profile.

Command Default

No default behavior or values.

Command Modes

Voice translation-profile configuration (cfg-translation-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use the tranlsate callback-number command to translate a called number to E.164 format. The translated number allows a called or calling number to be presented in its local form. The translate callback-number command is applied when translation-profile is configured on dialpeers, ephone-dn, and voice register-dn. The translate callback-number command is effective when the configruation setup reached the SCCP and SIP IP phones.

Examples

The following example shows a configuration in which a translation profile called name1 is created with two voice translation rules. Rule1 consists of associated calling numbers, and rule2 consists of redirected called numbers. The Cisco IP phones in SRST mode are configured with name1.

```
voice translation-profile name1
 translation calling rule1
 translation called-direct rule2
call-manager-fallback
 translation-profile incoming name1
```

Related Commands

Command	Description
show voice translation-profile	Displays the configuration of a translation profile.
translate (call-manager- fallback)	Applies a translation rule to modify the phone number dialed or received by any Cisco IP phone user during CallManager fallback.
translation-rule	Creates a translation name and enters translation-rule configuration mode to apply rules to the translation name.

Command	Description
voice translation-profile	Defines a translation profile for voice calls.

translate-outgoing (voice register pool)

To allow an explicit setting of translation rules on the VoIP dial peer in order to modify a phone number dialed by any Cisco IP phone user, use the **translate-outgoing** command in voice register pool configuration mode. To disable translation rules, use the **no** form of this command.

translate-outgoing {**called** | **calling**} *rule-tag*
no translate-outgoing {**called** | **calling**}

Syntax Description

called	Called party requires translation.
calling	Calling party requires translation.
rule-tag	The rule-tag is an arbitrarily chosen number by which the rule set is referenced. The range is from 1 to 2147483.

Command Default

Translation rules are enabled on the VoIP dial peer.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.

Usage Guidelines

Translation rules are a powerful general-purpose number-manipulation mechanism that perform operations such as automatically adding telephone area and prefix codes to dialed numbers. The translation rules are applied to VoIP dial peers created by Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME).

During registration, a dial peer is created, and that dial peer includes a default translation rule. The **translate-outgoing** command allows you to change the translation rule, if desired. The **translate-outgoing** command allows you to select a preconfigured number translation rule to modify the number dialed by a specific extension.

Translation rules must be set by using the **translate-outgoing** command before the **alias** command is configured in Cisco Unified SIP SRST.

Configure the **id** (voice register pool) command before any other voice register pool commands, including the **translate-outgoing** command. The **id** command identifies a locally available individual SIP phone or set of SIP phones.

Examples

Cisco Unified CME

The following is partial sample output from the **show running-config** command showing that called-party 1 requires translation.

```
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
```

Cisco Unified SIP SRST

The following is partial sample output from the **show running-config** command showing that called-party 1 requires translation.

```
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
 proxy 10.2.161.187 preference 1 monitor probe icmp-ping
 alias 1 94... to 91011 preference 8
 voice-class codec 1
```

Related Commands

	Description
alias (voice register pool)	Allows Cisco SIP IP phones to handle inbound PSTN calls to telephone numbers that are unavailable when the main proxy is not available.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
translate-outgoing (dial-peer)	Applies a translation rule to manipulate dialed digits on an outbound POTS or VoIP call leg.
voice register pool	Enters voice register pool configuration mode for SIP phones.

translation-profile

To assign a translation profile for incoming or outgoing call legs on a Cisco Unified IP phone, use the **translation-profile** command in ephone-dn or ephone-dn-template configuration mode. To delete the translation profile from the voice port, use the **no** form of this command.

translation-profile {**incoming** | **outgoing**} *name*
no translation-profile {**incoming** | **outgoing**} *name*

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
outgoing	Specifies that this translation profile handles outgoing calls.
name	Name of the translation profile.

Command Default

No default behavior or values

Command Modes

Ephone-dn configuration (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **translation-profile** command to assign a global predefined translation profile to an incoming or outgoing call leg.

If you use an ephone-dn template to apply a command to an ephone-dn and you also use the same command in ephone-dn configuration mode for the same ephone-dn, the value that you set in ephone-dn configuration mode has priority.

Examples

The following example assigns the translation profile named call_in to handle translation of incoming calls and a translation profile named call_out to handle outgoing calls:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2555
Router(config-ephone-dn)# translation-profile incoming call_in
Router(config-ephone-dn)# translation-profile outgoing call_out
```

The following example uses an ephone-dn-template to assign the translation profile named call_in to handle translation of incoming calls and the translation profile named call_out to handle outgoing calls:

```
Router(config)# ephone-dn-template 10
Router(config-ephone-dn-template)# translation-profile incoming call_in
Router(config-ephone-dn-template)# translation-profile outgoing call_out
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2555
Router(config-ephone-dn)# ephone-dn-template 10
```

Related Commands

	Description
show voice translation-profile	Displays the configuration of a translation profile.
translate	Applies a translation rule to modify the phone number dialed or received by any Cisco Unified IP phone user.
translation-rule	Creates a translation name and enters translation-rule configuration mode.
voice translation-profile	Defines a translation profile for voice calls.
voice translation-rule	Defines a translation rule for voice calls.

translation-profile incoming

To assign a translation profile for incoming call legs on a SIP phone, use the **translation-profile incoming** command in voice-register-dn configuration mode. To delete the translation profile from the directory number, use the **no** form of this command.

translation-profile incoming *name*
no translation-profile incoming

Syntax Description

name	Name of the translation profile to apply to incoming calls to this directory number. This is the *name* argument that was created for the profile with the **voice translation-profile** command.

Command Default

No translation profile is assigned to the directory number.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Use this command to assign a predefined translation profile to incoming call legs on the specified directory number. The translation profile that you assign is created by using the **voice translation-profile** command.

Examples

The following example shows that the translation profile named call_in is assigned to handle translation of incoming calls to directory number 1:

```
Router(config)# voice register dn 1
Router(config-register-dn)# number 2555
Router(config-register-dn)# translation-profile incoming call_in
```

Related Commands

	Description
show voice translation-profile	Displays the configuration of a translation profile.
translate (translation profiles)	Associates a translation rule with a voice translation profile.
voice translation-profile	Defines a translation profile for voice calls.
voice translation-rule	Defines a translation rule for voice calls.

transport (voice register pool-type)

To define the default transport type supported by the new phone, use the **transport** command in voice register pool-type mode. To remove the description, use the **no** form of this command.

Syntax Description

udp	(Optional) Selects UDP as the transport layer protocol. This is the default transport protocol.
tcp	(Optional) Selects TCP as the transport layer protocol.

Command Default

The default transport protocol is UDP. When the reference-pooltype command is configured, the transport value of the reference phone is inherited.

Command Modes

Voice Register Pool-Type Configuration (config-register-pooltype)

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to define the default transport type. If this parameter is not configured, UDP is used as default value. Currently, except the CiscoMobile-iOS and Jabber-Android, all other phone types uses UDP as default transport type. The default transport type will be ignored when the 'session-transport {udp | tcp}' command is configured for the pool.

Example

The following example shows how to specify a description for a phone model using the description command:

```
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# transport tcp
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.

trunk

To associate an ephone-dn with a foreign exchange office (FXO) port, use the **trunk** command in ephone-dn configuration mode. To disassociate the ephone-dn from the trunk number, use the **no** form of this command.

trunk *digit-string* [**timeout** *seconds*] [**transfer-timeout** *seconds*] [**monitor-port** *port*]
no trunk

Syntax Description

digit-string	The number of the trunk line.
timeout *seconds*	(Optional) Interdigit timeout between dialed digits, in seconds. Range is 3 to 30. Default is 3.
transfer-timeout *seconds*	(Optional) Number of seconds that Cisco Unified CME waits for the transfer-to party to answer a call after which the call is recalled to the phone that initiated the transfer. This keyword is supported for dual-line ephone-dns only. Range is 5 to 60000. Default is disabled.
monitor-port *port*	(Optional) Enables a button lamp or icon that shows that the specified port is in use. *Port* argument is platform-dependent; type **?** to display syntax.

Command Default

Ephone-dns are not associated with FXO ports.

Command Modes

Ephone-dn configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **monitor-port** and **transfer-timeout** keywords were added and support for dual-line ephone-dns was added.
12.4(9)T	Cisco Unified CME 4.0	This command with modifications was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to configure ephone-dns to support FXO lines that allow phones to have private lines connected directly to the PSTN. To bind the ephone-dn to the FXO port, use the destination pattern configured for the FXO line's POTS dial peer for the *digit-string* argument.

The **timeout** *seconds* argument controls the interdigit delay period, during which digits are collected from the user, and the delay before the connection to the FXO port is established. The argument controls the amount of time that Cisco Unified CME waits to collect digits for the dialed number, so that the digits can be included in the redial buffer and the Placed Calls directory of the phone. Digits that are entered after the timeout period are not included in the redial buffer or in the Placed Calls directory on the phone. The timeout parameter does not affect the time used to cut through the connection from the phone's trunk button to the FXO port. The phone user must either enter the pound (#) key or wait for this interdigit timeout to complete digit collection.

The phone user also has the option to use the phone's on-hook dialing feature so that the phone itself performs complete dial-string digit collection before signaling off-hook to the Cisco Unified CME. In this case all digits will be included in the Redial and Placed Calls Directory.

The **monitor-port** keyword enables direct status monitoring of the FXO port on the line button of the IP phone. The line button indicator, either a lamp or an icon depending on the phone, shows the in-use status of the FXO port during the duration of the call.

The **transfer-timeout** argument enables a transferred call to be automatically recalled if the transfer target does not answer after the specified number of seconds. The call is withdrawn from the transfer-to phone and the call resumes ringing on the phone that initiated the transfer.

The **monitor-port** and **transfer-timeout** keywords are not supported on ephone-dns for analog ports on the Cisco VG 224.

For dual-line ephone-dns, the second channel cannot receive incoming calls when the **trunk** command is configured.

Examples

The following example shows the configuration for two phones that each have a private FXO line button and a shared-line button.

The shared line’s voice ports and dial peers are as follows:

```
Router(config)# voice-port 1/0/1
Router(config-voice-port)# connection plar-opx 1000
Router(config)# dial-peer voice 101 pots
Router(config-dial-peer)# destination-pattern 9
Router(config-dial-peer)# port 1/0/1
The private lines’ voice ports and dial peers are as follows:
Router(config)# voice-port 1/1/0
Router(config-voice-port)# connection plar-opx 5550111
Router(config)# dial-peer voice 110 pots
Router(config-dial-peer)# destination-pattern 80
Router(config-dial-peer)# port 1/1/0
Router(config)# voice-port 1/1/1
Router(config-voice-port)# connection plar-opx 5550112
Router(config)# dial-peer voice 111 pots
Router(config-dial-peer)# destination-pattern 81
Router(config-dial-peer)# port 1/1/1
The following is the configuration for the shared and private ephone-dns:
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1000
Router(config-ephone-dn)# name Line1
Router(config-ephone-dn)# no huntstop
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 5550111
Router(config-ephone-dn)# name Private line
Router(config-ephone-dn)# trunk 80
Router(config)# ephone-dn 3
Router(config-ephone-dn)# number 5550112
Router(config-ephone-dn)# name Private line
Router(config-ephone-dn)# trunk 81
```

The following is the configuration for ephones with button 1 as a shared line and button 2 a private line:

```
Router(config)# ephone 1
Router(config-ephone)# mac-address 1111.1111.1101
Router(config-ephone)# button 1:1 2:2
Router(config)# ephone 2
Router(config-ephone)# mac-address 1111.1111.1102
```

```
Router(config-ephone)# button 1:1 2:3
```

The following example shows that transferred calls are recalled after 30 seconds if the destination party does not answer and status monitoring is enabled for FXO port 1/1/1.

```
Router(config)# ephone-dn 5
Router(config-ephone-dn)# trunk 801 timeout 5 transfer-timeout 30 monitor-port 1/1/1
```

Related Commands

	Description
destination-number	Specifies a connection mode for a voice port.

trustpoint (credentials)

To specify the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with the Cisco Unified SRST router certificate, use the **trustpoint** command in credentials configuration mode. To change the specified trustpoint, use the **no** form of this command.

trustpoint *trustpoint-name*
no trustpoint

Syntax Description

trustpoint-name	Name of the trustpoint to be associated with the Cisco Unified CME CTL provider certificate or the Cisco Unified SRST device certificate.

Command Default

No default behavior or values.

Command Modes

Credentials configuration (config-credentials)

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco SRST 3.3	This command was introduced for Cisco SRST.
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced for Cisco Unified CME.
12.4(9)T	Cisco Unified CME 4.0	This command for Cisco Unified CME was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Cisco Unified CME

This command is used with Cisco Unified CME phone authentication to define the trustpoint for the CTL provider. This trustpoint will be used for TLS sessions with the CTL client.

Cisco Unified SRST

The name of the trustpoint must be consistent with the trustpoint name of the Cisco Unified SRST router.

Examples

Cisco Unified CME

The following example sets up a CTL provider on the Cisco Unified CME router with the IP address 172.19.245.1.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint ctlpv
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```

Cisco Unified SRST

The following example enters credentials configuration mode, sets the IP source address and port, and specifies the trustpoint:

```
Router(config)# credentials
Router(config-credentials)# ip source-address 10.6.21.4 port 2445
Router(config-credentials)#
trustpoint srstca
```

Related Commands

	Description
ctl-service admin	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
debug credentials	Sets debugging on the credentials service.
ip source-address (credentials)	Enables the router to receive messages through the specified IP address and port.
show credentials	Displays the credentials settings.

trustpoint-label

To specify the PKI trustpoint label to be used for the TLS connection between the CAPF server and the phone, use the **trustpoint-label** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

trustpoint-label *label*
no trustpoint-label

Syntax Description

label	Trustpoint name for the CAPF server.

Command Default

No trustpoint label is specified for TLS connections.

Command Modes

CAPF-server configuration (config-capf-server)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to provide a PKI trustpoint name for the CAPF server. This trustpoint label is used for the TLS connection between the CAPF server and the phone.

Examples

The following example defines the CAPF server trustpoint name as server25.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

type

To assign a phone type to an SCCP phone, use the **type** command in ephone or ephone-template configuration mode. To remove a phone type, use the **no** form of this command.

type *phone-type* [**addon 1** *module-type* [**2** *module-type*]]
no type *phone-type* [**addon 1** *module-type* [**2** *module-type*]]

Syntax Description

phone-type	Type of phone. The following phone types are predefined in the system: • **12SP**—12SP+ and 30VIP phones. • 6901—Cisco Unified IP Phone 6901. • 6911—Cisco Unified IP Phone 6911. • 6921— Cisco Unified IP Phone 6921. • 6941—Cisco Unified IP Phone 6941. • **6945**—Cisco Unified IP Phone 6945. • 6961—Cisco Unified IP Phone 6961. • **7902**—Cisco Unified IP Phone 7902G. • **7905**—Cisco Unified IP Phone 7905G. • **7906**—Cisco Unified IP Phone 7906. • **7910**—Cisco Unified IP Phones 7910 and 7910G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco Unified IP Phone 7912G. • **7920**—Cisco Unified IP Phone 7920. • **7921**—Cisco Unified Wireless IP Phone 7921. • **7925**—Cisco Unified Wireless IP Phone 7925. • **7931**—Cisco Unified IP Phone 7931G. • **7935**—Cisco Unified IP Conference Station 7935. • **7936**—Cisco Unified IP Conference Station 7936. • **7937**—Cisco Unified IP Conference Station 7937. • **7940**—Cisco Unified IP Phone 7940G. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941G-GE.

	• **7942**—Cisco Unified IP Phone 7942. • **7945**—Cisco Unified IP Phone 7945 • **7960**—Cisco Unified IP Phone 7960G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961G-GE. • **7962**—Cisco Unified IP Phone 7962. • **7965**—Cisco Unified IP Phone 7965. • **7970—Cisco Unified IP Phone 7970G.** • **7971**—Cisco Unified IP Phone 7971G-GE. • **7975**—Cisco Unified IP Phone 7975. • **7985**—Cisco Unified IP Phone 7985. • **8941**—Cisco Unified IP Phone 8941. • **8945**—Cisco Unified IP Phone 8945. • **8961**—Cisco Unified IP Phone 8961. • **9951**—Cisco Unified IP Phone 9951. • **9971**—Cisco Unified IP Phone 9971. • **anl**—Analog. • **ata**—Cisco ATA-186 or Cisco ATA-188. • **bri** —SCCP Gateway (BR). • **vgc-phone** —VG248 phone emulation for analog phone. **Note** You can also add a new phone type to your configuration by using the **ephone-type** command.
addon 1 *module-type*	(Optional) Tells the router that an expansion module is being added to this Cisco Unified IP Phone and the type of module. Valid entries for *module-type* are: • **7914**—Cisco Unified IP Phone 7914 Expansion Module. • **7915-12**—Cisco Unified IP Phone 7915 12-Button Expansion Module. • **7915-24**—Cisco Unified IP Phone 7915 24-Button Expansion Module. • **7916-12**—Cisco Unified IP Phone 7916 12-Button Expansion Module. • **7916-24**—Cisco Unified IP Phone 7916 24-Button Expansion Module. **Note** This keyword is not supported for user-defined phone types created with the **ephone-type** command.
2 *module-type*	(Optional) Tells the router that a second expansion module is being added to this Cisco Unified IP Phone and the type of module. Valid entries for *module-type* are: • **7914**—Cisco Unified IP Phone 7914 Expansion Module. • **7915-12**—Cisco Unified IP Phone 7915 12-Button Expansion Module. • **7915-24**—Cisco Unified IP Phone 7915 24-Button Expansion Module. • **7916-12**—Cisco Unified IP Phone 7916 12-Button Expansion Module. • **7916-24**—Cisco Unified IP Phone 7916 24-Button Expansion Module. **Note** This keyword is not supported for user-defined phone types created with the **ephone-type** command.

Command Default

No phone type or add-on expansion module is defined.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The following keywords were added to this command: **7902**, **7905**, and **7912**.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2(1)	The **7970** keyword was added.
12.3(14)T	Cisco CME 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were added. This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	The **7931** keyword was added.
12.4(4)XC4	Cisco Unified CME 4.0(3)	The **7931** keyword was added.
12.4(11)T	Cisco Unified CME 4.0(3)	This command was integrated into Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **7921** and **7985** keywords were introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(15)T1	Cisco Unified CME 4.1(1)	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were introduced.
12.4(15)XZ	Cisco Unified CME 4.3	Support for user-defined phone types created with the **ephone-type** command was added.
12.4(15)XZ1	Cisco Unified CME 4.3	The **7915-12**, **7915-24**, **7916-12**, **7916-24**, and **7937** keywords were added.
12.4(20)T	Cisco Unified CME 7.0	The **7915-12**, **7915-24**, **7916-12**, **7916-24**, and **7937** keywords were added and this command was integrated into Cisco IOS Release 12.4(20)T.

Cisco IOS Release	Cisco Product	Modification
12.4(20)T1	Cisco Unified CME 7.0	The **7925** keyword was added.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. The 6921, 6941, 6961, and IP-STE keywords were added.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. The **6901** and **6911** keywords were added.
15.2(1)T	Cisco Unified CME 8.8	This command was modified. The **6945**, **8941**, and **8945** keywords were added.
15.3(3)M	Cisco Unified CME 10.0	This command was modified. The **7906**, **8961**, **9951**, and **9971** keywords were added.

Usage Guidelines

Not all phone types support add-on expansion modules. For support information, see User Documentation for Cisco Unified IP Phones.

This command must be followed by a phone reboot using the **reset** command.

If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example defines the IP phone with phone-tag 10 as a Cisco Unified IP Phone 7960G with two attached Cisco Unified IP Phone 7914 Expansion Modules:

```
Router(config)# ephone 10
Router(config-ephone)# type 7960 addon 1 7914 2 7914
```

The following example defines the IP phone with phone-tag 4 as a Cisco ATA device:

```
Router(config)# ephone 4
Router(config-ephone)# mac 1234.87655.234
Router(config-ephone)# type ata
```

The following example defines the IP phone with phone-tag 10 as a Cisco Unified IP Phone IP-STE:

```
Router(config)# ephone 10
Router(config-ephone)# type IPSTE
```

Related Commands

Command	Description
ephone-type	Adds a Cisco Unified IP phone type by defining a phone-type template.
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.

Command	Description
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.

type (voice register dialplan)

To specify a phone type for a SIP dial plan, use the **type** command in voice register dialplan configuration mode. To remove a phone type, use the **no** form of this command.

type *phone-type*
no type

Syntax Description

phone-type	Type of SIP phone for which the dial plan is used. Values are: • **7905-7912**—Cisco Unified IP Phone 7905, 7905G, 7912, or 7912G. • **7940-7960-others**—Cisco Unified IP Phone 7911, 7940, 7940G, 7941, 7942, 7941GE, 7945, 7960, 7960G, 7961, 7961GE, 7962, 7965, 7970, 7971, or 7975.

Command Default

The phone type is not defined.

Command Modes

Voice register dialplan configuration (config-register-dialplan)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(15)XZ	Cisco Unified CME 4.3	Support for Cisco Unified IP Phone 7942, 7945, 7962, 7965, and 7975 was added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies the type of SIP phone for which the dial plan is defined. You must use this command before defining dial patterns with the **pattern** command or selecting a dial pattern file in flash with the **filename** command.

The phone type specified with this command must match the phone type specified with the **type** command in voice register pool mode. If the dial plan type does not match the type assigned to the phone, the dial-plan configuration file is not generated.

Examples

The following example shows a SIP dial plan being defined for a Cisco Unified IP Phone 7905 or Cisco Unified IP Phone 7912:

```
Router(config)# voice register dialplan 10
Router(config-register-dialplan)# type 7905-7912
Router(config-register-dialplan)# pattern 52...
Router(config-register-dialplan)# pattern 91.......
```

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.
filename	Specifies a custom XML file that contains the dial patterns to use for the SIP dial plan.
pattern (voice register dialplan)	Defines a dial pattern for a SIP dial plan.
show voice register dialplan	Displays configuration information for a specific SIP dial plan.
type (voice register pool)	Defines a phone type for a SIP phone.

type (voice register pool)

To define a phone type for a SIP phone, use the **type** command in voice register pool configuration mode. To remove a phone type, use the **no** form of this command.

type *phone-type* [**addon 1 CKEM** | **CP-8800-Audio** | **CP-8800-Video**[**2 CKEM** | **CP-8800-Audio** | **CP-8800-Video**[**3 CKEM** | **CP-8800-Audio** | **CP-8800-Video**]]]
no type

Syntax Description

phone-type	

	Type of SIP phone that is being defined. Valid entries are as follows: • **3905**—Cisco Unified IP Phone 3905. • **3951**—Cisco Unified IP Phones 3911 and 3951. • **6901**—Cisco Unified IP Phone 6901. • **6911**—Cisco Unified IP Phone 6911. • **6921**—Cisco Unified IP Phone 6921. • **6922**—Cisco Unified IP Phone 6922. • **6941**—Cisco Unified IP Phone 6941. • **6945**—Cisco Unified IP Phone 6945. • **6961**—Cisco Unified IP Phone 6961. • **7821**—Cisco Unified IP Phones 7821. • **7841**—Cisco Unified IP Phones 7841. • **7861**—Cisco Unified IP Phones 7861. • **7905**—Cisco Unified IP Phones 7905 and 7905G. • **7906**—Cisco Unified IP Phone 7906G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco IP Phones 7912 and 7912G. • **7940**—Cisco IP Phones 7940 and 7940G. • **7941**—Cisco IP Phone 7941G. • **7941GE**—Cisco IP Phone 7941GE. • **7942**—Cisco Unified IP Phone 7942. • **7945**—Cisco Unified IP Phone 7945. • **7960**—Cisco IP Phones 7960 and 7960G. • **7961**—Cisco IP Phone 7961G. • **7961GE**—Cisco IP Phone 7961GE. • **7962**—Cisco Unified IP Phone 7962. • **7965**—Cisco Unified IP Phone 7965. • **7970**—Cisco IP Phone 7970G. • **7971**—Cisco IP Phone 7971GE. • **7975**—Cisco Unified IP Phone 7975. • **8851**—Cisco Unified IP Phone 8851. • **8851NR**—Cisco Unified IP Phone 8851NR. • **8861**—Cisco Unified IP Phone 8861. • **8865**—Cisco IP Phone 8865. • **8961**—Cisco Unified IP Phone 8961. • **9900**—Cisco Unified IP Phone 9900. • **9951**—Cisco Unified IP Phone 9951. • **9971**—Cisco Unified IP Phone 9971. • **ATA**—Cisco ATA-186 or Cisco ATA-188. • **ATA-187**—Cisco ATA-187. • **ATA-190**—Cisco ATA-190. • **ATA-191**—Cisco ATA-191. • **DX650**—Cisco DX650. • **Jabber-Android**—Cisco Jabber App on Android. • **P100**—PingTel Xpressa 100.

	• **P600**—Polycom SoundPoint 600. • **Jabber-CSF-Client**—Cisco Jabber CSF Client.
addon 1 CKEM	(Optional) Tells the router that a Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP phone. **Note** This option is available to Cisco Unified 8961, 9951, and 9971 SIP IP phones only.
2 CKEM	(Optional) Tells the router that a second Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP phone. **Note** This option is available to Cisco Unified 9951 and 9971 SIP IP phones only.
3 CKEM	(Optional) Tells the router that a third Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP phone. **Note** This option is available to Cisco Unified 9971 SIP IP phones only.
addon 1 CP-8800-Audio or **addon 1 CP-8800-Video**	(Optional) Tells the router that a Cisco SIP IP Phone 28-Button Line A-KEM or V-KEM is being added to this Cisco Unified SIP IP Phone.
2 CP-8800-Audio or **2 CP-8800-Video**	(Optional) Tells the router that a second Cisco SIP IP Phone A-KEM or V-KEM is being added to this Cisco Unified SIP IP Phone.
2 CP-8800-Audio or **2 CP-8800-Video**	(Optional) Tells the router that a second Cisco SIP IP Phone A-KEM or V-KEM is being added to this Cisco Unified SIP IP Phone.

Command Default

No phone type is defined.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was modified to add the **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970**, and **7971** keywords.
12.4(15)T	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970**, and **7971** keywords were integrated into Cisco IOS Release 12.4(15)T.
12.4(15)XZ	Cisco Unified CME 4.3	This command was modified to add the **7942**, **7945**, **7962**, **7965**, and **7975** keywords.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.1(3)T	Cisco Unified CME8.5	This command was modified to add the **8961**, **9951**,and **9971** keywords.

Cisco IOS Release	Cisco Product	Modification
15.2(1)T	Cisco Unified CME 8.8	This command was modified to add the **3905** keyword.
15.2(2)T	Cisco Unified CME 9.0	This command was modified to add the **6901**, **6911**, **6921**, **6941**, **6945**, **6961**, **ATA-187**, and **Jabber-Android** keywords.
15.2(4)M	Cisco Unified CME 9.1	This command was modified to include the **addon 1 CKEM**, **2 CKEM**, and **3 CKEM** keywords.
15.3(3)M	Cisco Unified CME 10.0	This command was modified to add the **6922** and **9900** keywords.
15.4(3)M	Cisco Unified CME 10.5	This command was modified. The **78XX**, **DX650** and **Jabber-CSF-Client**keywords were added.
Cisco IOS XE Gibraltar 16.10.1a Release	Unified CME 12.5	This command was modified. The **ATA-191**, **CP-8800-Audio**, and **CP-8800-Video**keywords were added.

Usage Guidelines

The **addon 1 CKEM**, **2 CKEM**, and **3 CKEM** keywords increase the number of speed-dial, busy-lamp-field, and directory number keys that can be configured.

There are two options in removing a Key Expansion Module (KEM) when you have configured all three KEMs.

The first option is to use the **no** form of the **type** command, then use the **type** command to configure only the KEMs to be included. The following example shows how the second and third KEMs are removed from the configuration:

```
Router(config)# voice register pool 9
Router(config-register-pool)# type 9971 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool)# no
 type 9971 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool)# type 9971 addon 1 CKEM
```

The second option is to define the same phone type while excluding from the configuration the KEM to be removed. For example, you have configured the following:

```
Router(config)# voice register pool 3
Router(config-register-pool)# type 9971 addon 1 CKEM 2 CKEM 3 CKEM
```

To remove the third KEM, enter the following:

Router(config-register-pool)# **type 9971 addon 1 CKEM 2 CKEM**

To remove the second KEM, enter the following:

```
Router(config-register-pool)# type 9971 addon 1 CKEM
```

From Unfiied CME 12.5 Release, **type** *phone-type* [**addon 1 CKEM** | **CP-8800-Audio** | **CP-8800-Video** [**2 CKEM** | **CP-8800-Audio** | **CP-8800-Video** [**3 CKEM** | **CP-8800-Audio** | **CP-8800-Video**]]] configuration is supported. The phone support is extended to Cisco IP Phone 8865 to support the Video KEM **CP-8800-Video**. Audio KEM **CP-8800-Audio** support is introduced for the Cisco IP Phone models 8851, 8851 NR, and 8861.

Note All the configuration characteristics of CKEM discussed here is applicable to **CP-8800-Audio** and **CP-8800-Video**.

After configuring the phone type, use the **create profile** command in voice register global configuration mode to generate the configuration profile files required for the phone and then reset or restart the phone using the **reset** or **restart** command, respectively.

Note Cisco Unified CME enables the **busy trigger-per-button (voice register pool)** command when phone-type **3905** is specified.

Examples

The following example shows how to define a SIP phone with phone-tag 10 as a Cisco Unified IP Phone 7960 or Cisco Unified IP Phone 7960G:

```
Router(config)# voice register pool 10
Router(config-register-pool)# type 7960
```

The following is a sample configuration for **CP-8800-Audio** and **CP-8800-Video** on the supported phone models for Unified CME 12.5 Release.

```
Router(config-register-pool)# type 8851 addon 1 CP-8800-Audio 2 CP-8800-Audio
Router(config-register-pool)# type 8851NR addon 1 CP-8800-Audio 2 CP-8800-Audio
Router(config-register-pool)# type 8861 addon 1 CP-8800-Audio 2 CP-8800-Audio 3 CP-8800-Audio
Router(config-register-pool)# type 8865 addon 1 CP-8800-Video 2 CP-8800-Video 3 CP-8800-Video
```

Related Commands

Command	Description
busy-trigger-per-button (voice register pool)	Sets the maximum number of calls allowed on a SIP directory number before activating Call Forward Busy or a busy tone.
load (voice register global)	Associates a type of Cisco Unified SIP IP phone with a phone firmware file.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.
reset (voice register pool)	Performs a complete reboot of one SIP phone associated with a Cisco Unified CME router.
restart (voice register)	Performs a fast reset of one or all SIP phones associated with a Cisco Unified CME router.
voice register pool	Enters voice register pool configuration mode for SIP phones.

type (voice-gateway)

To define the type of voice gateway to autoconfigure, use the **type** command in voice-gateway configuration mode. To remove the type from the configuration, use the **no** form of this command.

type {**vg202** | **vg204** | **vg224**}
no type

Syntax Description

vg202	Cisco VG202 Voice Gateway with 2 FXS ports.
vg204	Cisco VG204 Voice Gateway with 4 FXS ports.
vg224	Cisco VG224 Voice Gateway with 24 FXS ports.

Command Default

No type is defined for the voice gateway to be autoconfigured.

Command Modes

Voice-gateway configuration (config-voice-gateway)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command specifies the type of Cisco voice gateway for which you are creating an XML configuration file.

Examples

The following example shows a configuration for the Cisco VG224 voice gateway:

```
voice-gateway system  1
 network-locale FR
 type VG224
 mac-address 001F.A30F.8331
 voice-port 0-23
 create cnf-files
```

Related Commands

Command	Description
create cnf-files (voice-gateway)	Generates the XML configuration files that are required to autoconfigure the Cisco voice gateway.
mac-address (voice-gateway)	Defines the MAC address of the voice gateway to autoconfigure.
voice-port (voice-gateway)	Identifies the ports on the voice gateway that register to Cisco Unified CME.

Cisco Unified CME Commands: U

upa

To specify the audio file used for the unauthorized precedence announcement, use the **upa** command in voice MLPP configuration mode. To disable use of this audio file, use the **no** form of this command.

upa *audio-url*
no upa

Syntax Description

audio-url	Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.

Command Default

No announcement is played.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that plays to callers when they attempt to make a precedence call by using a higher level of precedence than the highest precedence level that is authorized for their line.

The **mlpp indication** command must be enabled (default) for a phone to play precedence announcements.

This command is not supported by Cisco IOS help. If you type **?** , Cisco IOS help does not display a list of valid entries.

Examples

The following example shows the unauthorized precedence announcement plays the file named upa.au located in flash:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# upa flash:upa.au
```

Related Commands

Command	Description
bnea	Specifies the audio file used for the busy station not equipped for preemption announcement.
ica	Specifies the audio file used for the isolated code announcement.
vca	Specifies the audio file used for the vacant code announcement.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.

upgrade (voice register global)

To generate a OS79XX.TXT file for firmware upgrades, use the **upgrade** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

upgrade
no upgrade

Syntax Description

This command has no arguments or keywords.

Command Default

No OS79XX.TXT file generated.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

The upgrade command performs the TFTP server alias binding, which can be verified with the **show voice register tftp-bind** command.

Examples

The following example shows the use of the **upgrade** command to upgrade Cisco SIP phone firmware from SIP [456].x to SIP [567].y with comments:

```
Router(config)# voice register global
Router(config-register-global)# load 7960-7940 P00x...
!Do not use file extension.
Router(config-register-global)# upgrade
!Generates OS79XX.txt file.
Router(config-register-global)# load 7960-7940 POSx...
!Do not use file extension. This
                                                       ! is only required if you
                                                       !are upgrading to 7.y.
Router(config-register-global)# create profile
!Generates SIPDefault.cnf and other files.
Router(config-register-global)# reset
Router(config-register-global)# no upgrade
!Returns  to default condition.
```

The P00x... and P0Sx... firmware filenames are required because the content in OS79XX.TXT is different from image_version tag in SIPDefault.cnf.

Related Commands

	Description
create profile (voice register global)	Generates configuration profile files required for SIP IP phones in Cisco Unified CME.
load (voice register global)	Associates a type of IP phone with a phone firmware file.

	Description
mode cme	Enables the mode for configuring SIP IP phones in Cisco Unified CME.
reset (voice register pool)	Reboots and reregisters a SIP IP phone, including contacting the DHCP server for updated information.
show voice register tftp-bind	Displays the current configuration files accessible to SIP phones.

url (telephony-service)

To provision uniform resource locators (URLs) for Cisco Unified IP phones connected to the Cisco Unified CME router, use the **url** command in telephony-service or group configuration mode. To remove a URL association, use the **no** form of this command.

url {**authentication** | **directories** | **idle** | **information** | **messages** | **proxy-server** | **services**} *url* [{*line* | **root**}]
no url {**authentication** | **directories** | **idle** | **information** | **messages** | **proxy-server** | **services**}

Syntax Description

authentication	Uses the information at the specified URL to validate requests made to the phone web server.
directories	Uses the information at the specified URL for the Directories button display.
idle	Information at the specified URL displays on the window of the IP phone during the idle state.
information	Uses the information at the specified URL for the Information button display. This button can be labeled "i" or "?". **Note** Cisco Unified CME does not support the use of this URL.
messages	Uses the information at the specified URL for the Messages button display.
proxy-server	Specifies the host and port used to enable proxy HTTP requests for access to remote host addresses from the phone HTTP client.
services	Uses the information at the specified URL for the Services button display.
url	URL as defined in RFC 2396.
line	(optional) Supported only with **services** keyword. Alphanumeric string of 1 to 32 characters that is line-services name to be displayed under Services button.
root	(optional) Supported only with **services** keyword and supported in telephony-service mode only. Menu of root phone services supported by a CSTA client application is displayed under Services button.

Command Default

The router automatically uses the local directory service.

Command Modes

Telephony-service configuration (config-telephony)
Group configuration (conf-tele-group)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was added to VRF group configuration mode.

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. Support for the **root** keyword was added to this command.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Cisco Unified IP Phones can support URLs in association with the four programmable feature buttons on those IP phones: Directories, Information, Messages, and Services. The fifth button, **Settings,** is managed entirely by the phone. Operation of these services is determined by the Cisco Unified IP phone capabilities and the content of the referenced URL.

This command provisions URLs through the configuration file supplied by the Cisco Unified CME router to the Cisco Unified IP phones during phone registration.

Note Cisco Unified CME does not support provisioning an information URL to access help using the i or ? buttons on a phone.

To use a Cisco Unified CallManager directory as an external directory source for Cisco Unified CME phones, the Cisco Unified CallManager must be made aware of the phones. You must list the MAC addresses of the Cisco Unified CME phones in the Cisco Unified CallManager and reset the phones from the Cisco Unified CallManager. It is not necessary for you to assign ephone-dns to the phones or for the phones to register with Cisco Unified CallManager.

Note Provisioning of the directory URL to select an external directory resource disables the Cisco Unified CME local directory service.

You can disable the local directory by using the **no service local-directory** command.

This command must be followed by a complete phone reboot using the **reset** command.

Examples

The following example provisions the Directories and Services buttons. Note that the Messages button is configured by the **voicemail** command. The Messages button acts like a speed-dial key to retrieve messages from a specified telephone number.

```
Router(config)# telephony-service
Router(config-telephony)# url directories http://1.4.212.11/localdirectory
Router(config-telephony)# url services http://1.4.212.4/CCMUser/123456/urltest.html
```

Related Commands

Command	Description
group	Creates a virtual router forwarding (VRF) group for Cisco Unified CME users and phones.
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.

Command	Description
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.
service local-directory	Enables the availability of the local directory service on IP phones.
voicemail	Defines the telephone number that is speed-dialed when the Messages button on a Cisco Unified IP phone is pressed.

url (voice register global)

To provision uniform resource locators (URLs) for feature buttons on Cisco SIP IP phones connected to a Cisco Unified CME router, use the **url** command in voice register global configuration mode. To remove a URL association, use the **no** form of this command.

url {**authentication** | **directory** | **service** | **idle**} *url*
no url {**authentication** | **directory** | **service** | **idle**}

Syntax Description

authentication	Uses the information at the specified URL to validate requests made to the phone web server.
directory	Uses the information at the specified URL for the Directories button display.
service	Uses the information at the specified URL for the Services button display.
idle	Uses the information at the specified URL to display on the IP phone during the idle state.
url	URL as defined in RFC 2396.

Command Default

The router automatically uses the local directory service.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.6.1	Cisco Unified CME 12.0	The **idle** keyword was added.
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

The Cisco Unified IP Phones 7940 and 7940G and Cisco Unified IP Phones 7960 and 7960G can support two URLs in association with two programmable feature buttons: Directories and Services. Operation of these services is determined by the Cisco IP phone capabilities and the content of the specified URL. The **Settings button** is managed entirely by the phone. The Messages button is configured by the **voicemail** command.

The purpose of the **url** command is to provision the URLs through the configuration file supplied by the Cisco Unified CME router to the SIP phones during phone registration.

You can disable the local directory by specifying the string "none" instead of a URL with the **directory** keyword, as shown in the following example:

```
Router(config)# voice register global
Router(config-register-global)# url directory none
```

Note Provisioning the directory URL to select an external directory resource disables Cisco Unified CME local directory service.

After you configure this command, restart the phone by using the **reset** command.

Examples

The following example shows how to provision the Directories and Services buttons:

```
Router(config)# voice register global
Router(config-register-global)# url directory http://1.4.212.11/localdirectory
Router(config-register-global)# url service http://1.4.212.4/CCMUser/123456/urltest.html
```

The following example shows that the information at the specified URL is used to validate requests made to the phone web server.

```
Router(config)# voice register global
Router(config-register-global)# url authentication http://CME IP
Address/CCMCIP/authenticate.asp
```

The following example specifies that the file logo.xml should be displayed on IP phones when they are not being used and that the display should be refreshed every 12 seconds:

```
Router(config)# voice register global
Router(config-register-global)# url idle http://mycompany.com/files/logo.xml idle-timeout
12
```

Related Commands

Commands	Description
reset (voice register pool)	Performs a complete reboot of one phone associated with a Cisco CME router.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
telephony-service	Enters telephony-service configuration mode.
voicemail (voice register template)	Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.

url (voice register template)

To define SIP phone URLs to configure dial rules such as Application Dial Rule, Directory Lookup Dial Rule, and LDAP server, use **url AppDialRule**, **url DirLookupRule**, and **url ldapServer** commands in voice register template configuration mode. To specify a file to display on an IP phone that is not in use, use the **url idle** command in voice register template configuration mode. To define a URL for invoking phone services, use the **url service** command in voice register template configuration mode.

url {**AppDialRule** | **DirLookupRule** | **ldapServer** | **idle** | **service**} {*string url*}

Syntax Description

url AppDialRule *string*	Application dial rule URL.
url DirLookupRule *string*	Directory lookup rule URL.
url ldapServer *string*	LDAP server URL.
url idle *url*	Defines the location of a file to display on phones that are not in use and specifies the interval between refreshes of the display, in seconds.
url service *url*	Uses the information at the specified URL for invoking phone services.

Command Default

No file is specified.

Command Modes

Voice register template configuration (config-register-template)

Command History

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.6.1	Unified CME 12.0	This command was modified to add **idle** and **service** keywords.

Usage Guidelines

Cisco softphone SIP client uses the dial rules to integrate with the Lightweight Directory Access Protocol (LDAP) directory server. The Cisco softphone SIP client also uses dial rules such as application dial rule and directory lookup rule to translate the outgoing phone numbers and display the incoming phone numbers with a rich caller ID. A rich caller ID displays a caller's name, caller's picture, or caller's phone number, or the information saved in the phone's directory.

You can create the application dial rule or directory lookup rule xml files and add these files to a tftp server. The Cisco softphone SIP client can download the dial rules using the **url ldapserver** *string*, **url AppDialRule** *string*, and **url DirLookupRule** *string* commands.

You can define the location of a file to display on phones that are not in use, and specify the interval between refreshes of the display using **url idle** command. You can also define a URL for invoking phone services using the **url service** command.

The following example shows how to define SIP phone URLs to configure Application Dial Rule, Directory Lookup Dial Rule, LDAP server, idle url, and service url in voice register template configuration mode.

```
Router(config-register-temp)# url ldapServer ldap.abcd.com
```

```
Router(config-register-temp)# url AppDialRule tftp://10.1.1.1/AppDialRules.xml

Router(config-register-temp)# url DirLookupRule tftp://10.1.1.1/DirLookupRules.xml

Router(config-register-temp)# url idle http://www.mycompany.com/files/logo.xml idle-timeout
 12

Router(config-register-temp)# url service http://10.0.0.4/CCMUser/123456/urltest.html
```

Related Commands

Commands	Description
voice register pool	Enters voice register pool configuration mode.
voice register template	Enters voice register template configuration mode.

url authentication

To instruct IP phones in Cisco Unified CME to send requests for authorization to a particular authentication server and include the specified credential in the requests, use the **url authentication** command in telephony-service configuration mode. To return to default, use the **no** form of this command.

url authentication *url-address application-name***password [0|6]** *password*
no url authentication *url-address application-name***password [0|6]** *password*

Syntax Description

url-address	URL adddress of authentication server. The URL address for the authentication server in Cisco Unified CME is: **http://***CME IP Address***/CCMCIP/authenticate.asp**.
application-name	Character string sent by application to identify itself to the server. Length of string: 1 to 15 characters. For applications other than Extension Mobility, the name portion of the credential must first be created in the application.
password [0|6]	Character string sent by application to identify itself to the server. Length of string: 1 to 15 characters. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption. For applications other than Extension Mobility, the password portion of the credential must first be created in the application.

Command Default

No authentication server or credential is specified for Cisco Unified CME to use for requesting authorization of commands from an application to a phone.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption, based on Unified CME password policy.

Usage Guidelines

This command specifies to which authentication server an IP phone in Cisco Unified CME must send requests for authorization and what credential to send in the request.

For Extension Mobility, use this command to instruct Extension Mobility phones to send an HTTP GET/POST to request authorization from the Cisco Unified CME authentication server before clearing call history when a user logs out.

For Extension Mobility, no additional commands are required to create or save the credential. The credential for the EM manager in Cisco Unified CME is whatever values you specifiy by using this command.

For applications other than Extension Mobility, the requisite credential must be created in the application.

To use the authentication server in Cisco Unified CME 4.3 and later versions to authorize requests for applications other than Extension Mobility, you must also configure the **authentication credential** command in telephony-service configuration mode.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]**. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Examples

The following example shows how to configure this command to instruct Extension Mobility phones in Cisco Unified CME to request authorization from the internal authentication server. The phones include the specified credential (extmob psswrd) in the requests.

```
Router(config)# telephony-service

Router(config-telephony)# url authentication http://192.0.2.0/CCMCIP/authenticate.asp extmob
 psswrd
Router(config-telephony)# exit
Router(config)#
```

Related Commands

Command	Description
authentication credential	Stores credentials in the database for the Cisco Unified CME authentication server.
keep call-history	Disables Automatic Clear Call History for Extension Mobility in Cisco Unified CME.

url idle

To specify a file to display on an IP phone that is not in use, use the **url idle** command in telephony-service configuration mode. To disable display of the file, use the **no** form of this command.

url idle *url* **idle-timeout** *seconds*
no url idle

Syntax Description

url	URL as defined in RFC 2396.
idle-timeout *seconds*	Time interval between display refreshes, in seconds. Range is from 0 to 300.

Command Default

No file is specified for display on idle phones.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

The file that is displayed must be encoded in eXtensible Markup Language (XML) using the Cisco XML document type definition (DTD). For more information about Cisco DTD formats, refer to Cisco IP Phone Services Application Development Notes.

This command must be followed by a complete phone reboot using the **reset** command.

Examples

The following example specifies that the file logo.xml should be displayed on IP phones when they are not being used and that the display should be refreshed every 12 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# url idle http://mycompany.com/files/logo.xml idle-timeout 12
```

Related Commands

	Description
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco CME router.

url services (ephone-template)

To provision up to eight uniform resource locators (URLs) for the Services feature button on individual SCCP phones connected to Cisco Unified CME, use the **url services** command in ephone-template configuration mode. To reset to the default, use the **no** form of this command.

url services *url-tag url url-name*
no url services *url-tag*

Syntax Description

url-tag	Identifier for url being configured for Services feature button. Range is 1 to 8.
url	URL as defined in RFC 2396.
url-name	Alpha-numerical string to appear for this URL in Services feature button display. Length of string is 1 to 256 contiguous characters (a-z, 0-9).

Command Default

The system-level configuration for the Services button is used.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command displays the information at up to eight URLs for the Services feature button display on a supported Cisco Unified IP phone. Operation of these services is determined by the capabilities of the Cisco Unified IP phone and the content of the specified URL.

If you use an ephone template to apply this command to one or SCCP phones and you also use the **url** command in telephony-service configuration mode to configure a services url for all SCCP phones, the value set in telephony-service configuration mode appears first in the list of options displayed when the phone user presses the Services feature button, before any URLs configured by using this command. Cisco Unified CME self-hosted services, such as Extension Mobility, always appears last in the list of options displayed for the Services feature button.

The number of *url-name* characters that appear on the IP phone display is not fixed because IP phones typically use a proportional font.

After creating an ephone template that contains a services URL, use the **ephone-template (ephone)** command to apply the template to individual phones.

Examples

The following example defines three urls for the Services feature button display, one for all SCCP phones and two others in an ephone-template that is applied to individual phones. Phones to which the template is applied (ephones 11 and 13) will have a second and third option in the Services feature button display.

```
telephony-service
 url services http://10.0.0.4/CMEUser/123456/urlsupport.html
.
.
.
 create cnf-files
.
.
.
.
ephone-template 1
 url services 1 http://10.0.0.4/CMEUser/123456/cal.html Calendar
 url services 2 http://10.0.0.4/CMEUser/123456/quotes.html StockQuotes
ephone 11
 mac-address F00D.EDAB.1234
 type 7960
 button 1:25
 ephone-template 1
ephone 12
 mac-address 0003.B0D5.6541
 type 7960
 button 1:26
 logout-profile 1
ephone 13
 mac-address 000D.3666.3D00
 type 7960
 ephone-template 1
 logout-profile 1
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an SCCP phone.
url (telephony-service)	Provisions URLs for programmable feature buttons on supported Cisco Unified IP phones.

url-button

To configure service url feature on a line key, use the **url-button** command in ephone-template mode. To unconfigure the service url feature on a line key, use the **no** form of this command.

url-button *index*{ **type** | **url** [**name**]}
no url-button *index* **typeurl** [**name**]

Syntax Description

index	Unique index number. The range is from 1 to 8.
type	Type of service URL button. The following types of URL service buttons are available: • myphoneapp: My phone application configured under phone user interface. • em: Extension Mobility • snr: Single Number Reach • voicehuntgroups: Displays a list of voice hunt groups • park-list: Displays a list of parked calls
url-button	Service url with maximum length of 31 characters.

Command Default

By default, URL-button configuration on a line key is not configured.

Command Modes

ephone template (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.
15.4(3)M	Cisco Unified CME 10.5	This command was modified to add voice hunt groups and park-list as new types .

Usage Guidelines

You can configure url-button feature on a line key to function as an extension mobility (EM), My Phone Apps, or single number reach (SNR). You can also configure the url-button feature on a line button to function as a service URL. by configuring a URL name of a maximum length of 31 characters.

Examples

The following examples shows three URL buttons configured on a line key:

```
!
telephony-service
 max-ephones 25
 max-conferences 12 gain -6
 transfer-system full-consult
!
!
ephone-template  5
 url-button 1 em
 url-button 2 mphoneapp
 url-button 3 snr
 url-button 4 voicehuntgroups
 url-button 5 park-list
!
```

```
ephone-template 6
conference drop-mode never
conference add-mode all
conference admin: No
max-calls-per-button 8
busy-trigger-per-button 0
privacy default
url-button 1 em
url-button 2 www.cisco.com www.cisco.com
url-button 3 snr
url-button 4 help help
url-button 7 myphoneapp
!
!
```

Related Commands

Command	Description
show telephony-service ephone-template	Displays the contents of all the ephone templates defined.

url-button (voice-register-template)

To configure service url feature button on a line key, use the url-button command in voice register template mode. To disable the service url feature button configuration on a line key, use the no form of this command.

url-button [**index number**] [{**url location** | **url name**}]
no url-button [**index number**] [{**url location** | **url name**}]

Syntax Description

index number	Unique index number. Range: 1 to 8.
url location	Location of the url.
url name	Service url with maximum length of 31 characters.

Command Default

URL-button configuration on a line key is disabled.

Command Modes

Voice register template configuration (config-register-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to configure a url-button on a phone's line key.You can configure a line key to function as a url-button. You can also configure a line button to function as a service url by configuring a url name of a maximum length of 31 characters.

Examples

The following example shows url-button configured in voice register template 1:

```
Router# show run
!
!
voice register template  1
 url-button 1 http://www.cisco.com cisco
 button-layout 1 line
 button-layout 2,5 speed-dial
!
voice register pool  50
 id mac 001E.7AC4.DC73
 feature-button 1 NewCall
 type 7965
 number 1 dn 65
 template 1
 dtmf-relay rtp-nte
 speed-dial 1 2001 label "SD1-2001"
```

Related Commands

Command	Description
show voice register pool	**Displays all configuration information associated with a particular voice register pool.**
show voice register template	Displays all configuration information associated with a SIP phone template,

user (voice logout-profile)

To create an authentication credential for use by Telephone Application Programming Interface (TAPI) phone devices and certain other applications to log into Cisco Unified CME, use the **username** command in voice logout-profile configuration mode. To remove the credential, use the **no** form of this command.

user *username* **password** **[0|6]** *password*
no **user** *name* **password** **[0|6]** *password*

Syntax Description

name	Unique alphanumeric string to be used by a TAPI phone device to log into Cisco Unified CME. String can contain a maximum of 15 alphanumeric characters.
password	Password to be used with this username for authentication purposes.
password [0|6]	Alphanumeric string. The 0 in the parameter [0|6] represents plain, unencrypted text and 6 represents level 6 password encryption.

Command Default

No authentication credential is created.

Command Modes

Voice logout-profile configuration (voice-logout-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption, based on Unified CME password policy.

Usage Guidelines

Use this command in voice logout-profile configuration mode to add an authentication credential to a logout profile for Extension Mobility. The authentication credential is used by TAPI phone devices and certain other applications to log into Cisco Unified CME via an IP phone that is enabled for Extension Mobility and on which the logout profile is downloaded.

The user name parameter of any authentication credential must be unique. Do not use the same value for a user name when you configure any two or more authentication credentials in Cisco Unified CME, such as the username for any Cisco United CME GUI account and the user name in a profile for Extension Mobility.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]**. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Examples

The following example shows the configuration for a logout profile that defines the default appearance for a Cisco Unified IP phone that is enabled for Extension Mobility, including the username (23C2-8) and password (43214) to be used by a TAPI phone device to log into Cisco Unified CME:

```
pin 9999
user 23C2-8 password 43214
number 3001 type silent-ring
number 3002 type beep-ring
number 3003 type feature-ring
number 3004 type monitor-ring
number 3005,3006 type overlay
number 3007,3008 type cw-overly
speed-dial 1 2000
speed-dial 2 2001 blf
```

Related Commands

Command	Description
logout-profile	Enables a Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones to which a particular logout-profile or user-profile is downloaded.

user (voice user-profile)

To create an authentication credential to be used by Extension Mobility in Cisco Unified CME, use the username command in voice user-profile configuration mode. To remove the credential, use the **no** form of this command.

user *name* **password** *password*
no **user** *name* **password** *password*

name	Unique alphanumeric string to identify a user for this authentication credential only. String can contain a maximum of 15 alphanumeric characters.
password	Password to be used with this user name for authentication purposes.
password	Alphanumeric string.

Command Default

Credential is not created.

Command Modes

Voice user-profile configuration (config-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command in voice user-profile configuration mode creates a credential to be authenticated by Cisco Unified CME before a phone user can log into a Cisco Unified IP phone that is enabled for Extension Mobility

The user name parameter of any authentication credential must be unique. Do not use the same value for a user name when you configure any two or more authentication credentials in Cisco Unified CME, such as the username for any Cisco United CME GUI account and the user name in a profile for Extension Mobility.

When a user logs into an extension mobility enabled phone, Cisco Unified CME retrieves the appropriate user profile, based on username and password match, and replace the phone's default logout profile with the user's profile.

Examples

The following example shows the configuration to be downloaded after a user enters the username and password configured in this profile, and Cisco Unified CME matches the entry to the credentials in a user profile database:

```
voice user-profile 1
 pin  12345
 user me password pass123
 number 2001 type silent-ring
 number 2002 type beep-ring
```

```
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

Command	Description
reset (voice logout-profile and voice user-profile)	Preforms complete reboot of all IP phones on which a particular logout-profile or user-profile is downloaded.

user-locale (ephone-template)

To specify a user locale in an ephone template, use the **user-locale** command in ephone-template configuration mode. To reset to the default user locale, use the **no** form of this command.

user-locale *user-locale-tag*
no user-locale

Syntax Description

user-locale-tag	Locale identifier that was assigned to the user locale using the **user-locale (telephony-service)** command.

Command Default

The default user locale (user-locale 0) is used.

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

To apply user locales to individual ephones, you must specify per-phone configuration files using the **cnf-file perphone** command and identify the locales using the **user-locale (telephony-service)** command.

After creating an ephone template that contains a locale tag, use the **ephone-template (ephone)** command to apply the template to individual ephones.

Examples

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 create cnf-files
ephone-template 1
 user-locale 1
 network-locale 1
ephone-template 2
 user-locale 2
 network-locale 2
ephone-template 3
 user-locale 3
 network-locale 3
ephone 11
```

```
 button 1:25
 ephone-template 1
ephone 12
 button 1:26
 ephone-template 2
ephone 13
 button 1:27
 ephone-template 3
ephone 14
 button 1:28
```

Related Commands

	Description
cnf-file	Specifies the type of configuration files that phones use.
ephone-template (ephone)	Applies an ephone template to an ephone.
user-locale (telephony-service)	Sets the language for displays on supported phone types.

user-locale (telephony-service)

To define languages for displays on supported phones, use the **user-locale** command in telephony-service configuration mode. To remove a locale configuration, use the **no** form of this command.

user-locale [*user-locale-tag*] [*user-defined-code*] *country-code* [**load** *TAR-filename*]
no user-locale [*user-locale-tag*] *country-code*

Syntax Description

user-locale-tag	(Optional) Identifier for the specified locale. Required to configure multiple locales only. Range is 0 to 4. Default is 0.
user-defined-code	(Optional) Label for locale that is not one of the 12 standard ISO 366 locales. Use each label for only one *user-locale-tag* at a time. Values are **U1**, **U2**, **U3**, **U4**, and **U5**.
country-code	• **DE**—Germany • **DK**—Denmark • **ES**—Spain • **FR**—France • **IT**—Italy • **JP**—Japan • **NL**—Netherlands • **NO**—Norway • **PT**—Portual • **RU**—Russia • **SE**—Sweden • **US**—United States • Any valid ISO 639 code to be associated with the *user-defined-code* argument (U1 to U5) only. Code must be for a supported locale that is not listed above and for which the XML files can be downloaded to flash, slot 0, or a TFTP server. • **U1**, **U2**, **U3**, **U4**, **U5**—Only when used with the **load** keyword and where U1 to U5 corresponds to a user-defined locale for which the TAR file is downloaded to flash, slot 0, or a TFTP server.
load	(Optional) Extracts contents of a TAR file to the location specified by using the **cnf-file location** command. This keyword is supported in Cisco Unified CME 7.0(1) and later versions.
TAR-filename	TAR file containing the language JAR file and the tg3-tones.xml file for country-specific network tones and cadences.

Command Default

The default user-locale tag is 0 and the default locale is US (United States).

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.

Cisco IOS Release	Cisco Product	Modification
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The following keywords were added: **DK**, **NL**, **NO**, **PT**, **RU**, and **SE**.
12.3(4)T	Cisco CME 3.0	The keywords added for Cisco CME 3.0 were integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	The *user-locale-tag* and *user-defined-code* arguments were added.
12.4(9)T	Cisco Unified CME 4.0	The *user-locale-tag* and *user-defined-code* arguments were integrated into Cisco IOS Release 12.4(9)T.
12.4(20)YA	Cisco Unified CME 7.0(1)	The **load** *TAR-filename* keyword/argument combination for the locale installer was added.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command sets the language for displays on supported phone types.

The **show telephony-service tftp-bindings** command displays the locale that is set using this command. This locale is associated with the dictionary and language files.

Follow this command with a complete phone reboot using the **reset** command.

User-locale 0 always holds the default language that is used for all SCCP phones that are not assigned alternative user locales or user-defined user locales. The system default is US (United States) unless you use this command to designate a different locale for user-locale 0.

Alternative User Locales

In Cisco Unified CME 4.0 or a later version, the *user-locale-tag* argument allows you to specify up to five alternative user locales. For example, a company can specify French for phones A, B, and C; German for phones D, E, and F; and United States for phones G, H, and I.

Each of the five user locales that you can use in a multi locale system is identified with the *user-locale-tag* argument. The identifier 0 always holds the default locale, although you can define this default to be any language code that is supported in the system and is listed in CLI help for the command. For example, if you define locale-tag 0 to be JP (Japanese), the default user locale for the router is JP. If you do not specify a locale for identifier 0, the default is US (United States). If you are using this command to configure a default locale for all SCCP phones in your system, you are not required to include *user-locale-tag 0* in the command.

To apply alternative user locales to different phones, you must use the **cnf-files** command to specify per-phone configuration files. When you use per-phone configuration files, a phone's configuration file automatically uses the default locales in user locale 0 and network locale 0. You can override this default for individual ephones by assigning locale tags to the alternative language codes that you want to use. Use ephone templates to assign the locale tag to individual ephones. For example, you can assign user-locale-tag 2 to the language code RU (Russian).

Use the **user-locale** command in ephone-template mode to apply a locale tag to an ephone template. Use the **ephone-template** command in ephone configuration mode to apply the template a phones that should use the alternative language.

User-Defined User Locales

In Cisco Unified CME 4.0 and later versions, you can install XML files to support up to five user and network locales that are not standard in your system. These files cannot be installed in the system storage location. To support user-defined locales, you must use the **cnf-files perphone** and **cnf-file location** commands and copy the appropriate XML language files into slot 0, flash, or TFTP memory. The user locales and network locales that are stored in this way can then be used as default or alternative entries for all or some phones.

For example, if you have a site at which the phones should use the displays and tones for Traditional Chinese, which is not one of the standard languages, you must download the XML files for Traditional Chinese to use this user-defined locale on a phone.

Locale Installer

In Cisco Unified CME 7.0(1) and later versions, this command with the **load** keyword is a locale installer that extracts the contents of the locale TAR file to the location specified by the **cnf-file location** command. Before Cisco Unified CME 7.0(1), you had to manually extract the files to flash, slot 0, or an external TFTP server.

Before using this command as a locale installer, you must manually create the required locale folders in the root directory of the external TFTP server.

Examples

The following example sets the default language tag for the IP phone display to French:

```
telephony-service
 user-locale FR
```

The following example sets the default language tag for the IP phone display to French. It shows another way to change the default:

```
telephony-service
 user-locale 0 FR
```

The following example sets the alternative language tag 1 to German:

```
telephony-service
 user-locale 1 DE
```

Cisco Unified CME 4.0 and Later Versions: Alternative User Locale

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
```

```
 user-locale 3 ES
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 create cnf-files
ephone-template 1
 user-locale 1
 network-locale 1
ephone-template 2
 user-locale 2
 network-locale 2
ephone-template 3
 user-locale 3
 network-locale 3
ephone 11
 button 1:25
 ephone-template 1
ephone 12
 button 1:26
 ephone-template 2
ephone 13
 button 1:27
 ephone-template 3
ephone 14
 button 1:28
```

Cisco Unified CME 4.0 and Later Versions: User-Defined User Locale

The following example applies locale tag 4 to the user-defined code U1, which is defined as ZH. ZH is the code that represents Traditional Chinese in ISO 639, the *Language Code Reference* . Because the code for Traditional Chinese is not one of those that is provided in the system, the user must download the appropriate XML files to support this language.

In addition to the user-defined code, the example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales; ephone 12 uses FR; ephone 13 uses ES; ephone 14 uses the default, US; and ephone 15 uses the user-defined language, ZH (Traditional Chinese).

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 user-locale 4 U1 ZH
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 network-locale 4 U1 ZH
 create cnf-files
ephone-template 1
 user-locale 1
 network-locale 1
ephone-template 2
 user-locale 2
 network-locale 2
ephone-template 3
 user-locale 3
```

```
 network-locale 3
ephone-template 4
 user-locale 4
 network-locale 4
ephone 11
 button 1:25
 ephone-template 1
ephone 12
 button 1:26
 ephone-template 2
ephone 13
 button 1:27
 ephone-template 3
ephone 14
 button 1:28
ephone 15
 button 1:29
 ephone-template 4
```

Cisco Unified CME 7.0(1) and Later Versions: Using Locale Installer

The following example is the output from the **user-locale** command when the user-defined locale is on the default locale index (0) and the country-code is U2 for user-defined Finnish. The contents of the TAR file are extracted to flash, slot 0, or a TFTP server as previously specified by the **cnf-file location** command.

```
Router(config-telephone)# user-locale U2 load CME-locale-fi_FI-7.0.1.1.tar
Updating CNF files
LOCALE INSTALLER MESSAGE: VER:1
LOCALE INSTALLER MESSAGE: Langcode:fi
LOCALE INSTALLER MESSAGE: Language:Finnish
LOCALE INSTALLER MESSAGE: Filename: 7905-dictionary.xml
LOCALE INSTALLER MESSAGE: Filename: 7905-kate.xml
LOCALE INSTALLER MESSAGE: Filename: 7920-dictionary.xml
LOCALE INSTALLER MESSAGE: Filename: 7960-dictionary.xml
LOCALE INSTALLER MESSAGE: Filename: 7960-font.xml
LOCALE INSTALLER MESSAGE: Filename: 7960-kate.xml
LOCALE INSTALLER MESSAGE: Filename: 7960-tones.xml
LOCALE INSTALLER MESSAGE: Filename: mk-sccp.jar
LOCALE INSTALLER MESSAGE: Filename: tc-sccp.jar
LOCALE INSTALLER MESSAGE: Filename: td-sccp.jar
LOCALE INSTALLER MESSAGE: Filename: tags_file
LOCALE INSTALLER MESSAGE: Filename: utf8_tags_file
LOCALE INSTALLER MESSAGE: Filename: g3-tones.xml
LOCALE INSTALLER MESSAGE: Filename: SCCP-dictionary.utf-8.xml
LOCALE INSTALLER MESSAGE: Filename: SCCP-dictionary.xml
LOCALE INSTALLER MESSAGE: Filename: ipc-sccp.jar
LOCALE INSTALLER MESSAGE: Filename: gp-sccp.jar
LOCALE INSTALLER MESSAGE: New Locale configured
Processing file:flash:/its/user_define_2_tags_file
Processing file:flash:/its/user_define_2_utf8_tags_file
CNF-FILES: Clock is not set or synchronized, retaining old versionStamps
CNF files updating complete

Router(config-telephony)# create cnf-files
Router(config-telephony)# ephone 3
Router(config-ephone)# reset
```

Related Commands

Command	Description
cnf-file location	Specifies a storage location for XML configuration files.
cnf-files	Specifies the type of phone configuration files to be created.
ephone-template (ephone)	Applies an ephone template to an ephone.
network-locale (telephony-service)	Selects a code for a geographically specific set of tones and cadences on supported phone types.
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service tftp-bindings	Displays the current configuration files that are accessible by IP phones.
user-locale (ephone-template)	Applies a user locale tag to an ephone template.

user-locale (voice register)

To define languages for display on supported Cisco Unified SIP IP phones, use the **user-locale** command in voice register global or voice register template configuration mode. To remove a locale configuration, use the no form of this command.

user-locale *[user-locale-tag]*{*[user-defined-code] country-code*}**[load TAR-***filename***]**

no user-locale *[user-locale-tag]*{*[user-defined-code] country-code*}**[load TAR-***filename***]**

Syntax Description

user-locale-tag	(Optional) Identifier for the specified locale. Required to configure multiple locales only. Range is 0 to 4. Default is 0.
user-defined-code	(Optional) Label for locale that is not one of the 12 standard ISO 366 locales. Use each label for only one user-locale-tag at a time. Values are **U1**, **U2**, **U3**, **U4**, and **U5**. This option is only available after the **tftp-path** command is configured in voice register global configuration mode and the directory in which the configuration files are written is specified (flash, slot, or an external TFTP server).
country-code	• **DE**—Germany • **DK**—Denmark • **ES**—Spain • **FR**—France • **IT**—Italy • **JP**—Japan • **NL**—Netherlands • **NO**—Norway • **PT**—Portugal • RU—Russia • **SE**—Sweden • U1—User defined user-locale 1 • **U2**—User defined user-locale 2 • **U3**—User defined user-locale 3 • **U4**—User defined user-locale 4 • **U5**—User defined user-locale 5 • **US**—United States • Any valid ISO 639 code to be associated with the user-defined-code argument (U1 to U5) only. Code must be for a supported locale that is not listed above and for which the XML files can be downloaded to flash, slot 0, or a TFTP server. • **U1**, **U2**, **U3**, **U4**, **U5**—Only when U1 to U5 corresponds to a user-defined locale for which the TAR file is downloaded to flash, slot 0, or a TFTP server.
load *TAR-filename*	(Optional) Loads the specified localization package file. **Note** Use the complete filename, including the file suffix (.tar), when you configure the user-locale command for all Cisco Unified SIP IP phone types.

Command Default

The default user-locale tag is 0 and the default locale is US (United States).

Command Modes

Voice register global (config-register-global)
Voice register template (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.
15.2(2)T	Cisco Unified CME 9.0	This command was introduced.

Usage Guidelines

This command sets the language for displays on supported phone types.

The **show voice register global** command displays the language (locale) that is set using this command. This locale is associated with the dictionary and language files.

Follow this command with a complete phone reboot using the**reset (voice register global)** command.

User-locale 0 always holds the default language that is used for all SIP phones that are not assigned alternative user locales or user-defined user locales. The system default is US (United States) unless you use this command to designate a different locale for user-locale 0.

Alternative User Locales

The user-locale-tag argument allows you to specify up to five alternative user locales. For example, a company can specify French for phones A, B, and C; German for phones D, E, and F; and United States for phones G, H, and I.

Each of the five user locales that you can use in a multilocale system is identified with the user-locale-tag argument. The identifier 0 always holds the default locale, although you can define this default to be any language code that is supported in the system and is listed in CLI help for the command. For example, if you define locale-tag 0 to be JP (Japanese), the default user locale for the router is JP. If you do not specify a locale for identifier 0, the default is US (United States). If you are using this command to configure a default locale for all SIP phones in your system, you are not required to include user-locale-tag 0 in the command.

Use the **user-locale** command in voice register template configuration mode to apply a locale tag to a voice register template. Use the **voice register template** command in global configuration mode to apply the template to phones that should use the alternative language.

Example

The following example sets the default language tag for the IP phone display to French:

```
voice register global
user-locale 0 FR
```

The following example sets alternative language tag 2 as CH (Chinese):

```
Tftp path is flash:
Generate text file is disabled
Tftp files are created, current syncinfo 0202310605309206
OS79XX.TXT is not created
timeout interdigit 10
network-locale[0] US (This is the default network locale for this box)
network-locale[1] US
network-locale[2] US
network-locale[3] US
network-locale[4] US
```

```
user-locale[0] U2 language code CH (This is the default user locale for this
box)
user-locale[1] US
user-locale[2] US
user-locale[3] US
user-locale[4] US Active registrations : 2
Total SIP phones registered: 2
Total Registration Statistics
Registration requests : 4
```

The following example sets user-locale 2 and 3 for voice register template 5 and 6, respectively:

```
voice register template 1
softkeys hold Resume Newcall
softkeys idle Redial DND Gpickup Pickup Cfwdall
softkeys connected Endcall Hold Confrn Park Trnsfer
softkeys remote-in-use Barge Newcall cBarge
no transfer-blind enable
!
voice register template 5
user-locale 2
!
voice register template 6
user-locale 3
!
```

The following example loads the locale package file for Germany:

```
Router(config)# voice register global
Router(config-register-global)# user-locale 2 DE load CME-locale-de_DE-German-8.6.3.0.tar
```

The following example loads the locale package file for Italy:

```
Router(config)#voice register global
Router(config-register-global)# user-locale IT load CME-locale-it_IT-Italian-8.6.2.4.tar

LOCALE INSTALLER MESSAGE (SIP):Loading Locale Package...
LOCALE INSTALLER MESSAGE: VER:2
LOCALE INSTALLER MESSAGE: Langcode:it_IT
LOCALE INSTALLER MESSAGE: Language:Italian
LOCALE INSTALLER MESSAGE: Filename: g3-tones.xml
LOCALE INSTALLER MESSAGE: Filename: tags_file
LOCALE INSTALLER MESSAGE: Filename: utf8_tags_file
LOCALE INSTALLER MESSAGE: Filename: gd-sip.jar
LOCALE INSTALLER MESSAGE: Filename: g4-tones.xml
LOCALE INSTALLER MESSAGE: New Locale configured
```

Related Commands

Command	Description
reset (voice register global)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.
show voice register global	Displays the current configuration files that are accessible to the Cisco Unified SIP IP phones.
voice register global	Sets global parameters for all supported Cisco SIP IP phones in a Cisco Unified CME environment.
voice register template	Defines a template of common parameters for Cisco Unified SIP IP phones.

username (ephone)

To assign a login account username and password to a phone user so that the user can log in to the Cisco Unified CME router through a web browser, use the **username** command in ephone configuration mode. To disable the username and password, use the **no** form of this command.

username *username* **password** **[0|6]** *password*
no **username** *username* **password** **[0|6]** *password*

Syntax Description

username	Unique alphanumeric string to identify a user for this authentication credential only. String can contain a maximum of 28 alphanumeric characters. Default is Admin.
password	Enables password for the Cisco Unified IP phone user.
password	Password string.

Command Default

The default username for the administrator is Admin.

Command Modes

Ephone configuration (config-ephone)

Command History

Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.1	This command was introduced.
12.2(8)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(8)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption, based on Unified CME password policy.

Usage Guidelines

This command assigns a login account username and password for a phone user and establishes a login account for each Cisco Unified IP phone. This configuration can be completed only by the local system administrator of the Cisco Unified CME router.

You must also create a login account to allow Telephone Application Programming Interface (TAPI)-aware PC applications to register with the Cisco router and exercise remote-control operation of a Cisco Unified IP phone.

The user name parameter of any authentication credential must be unique. Do not use the same value for a user name when you configure any two or more authentication credentials in Cisco Unified CME, such as the username for any Cisco United CME GUI account and the user name in a profile for Extension Mobility.

This configuration permits the phone user to log in to Cisco Unified CME to view and change attributes associated only with the user's IP phone.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]**. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Examples

The following example shows how to set the username and password:

```
Router(config)# ephone 1
Router(config-ephone)# username smith password 9golf
```

Related Commands

	Description
admin-password	Sets a password for the local system administrator of the Cisco IOS Telephony Service.
admin-username	Sets the username for the local system administrator of the Cisco IOS Telephony Service router.

username (voice register pool)

To assign an authentication credential to a phone user so that the SIP phone can register in Cisco CallManager Express (Cisco CME), use the **username** command in voice register pool configuration mode. To disable a username and password, use the **no** form of this command.

username *username* [**password** **[0|6]** *password*]
no **username** *username* [**password** **[0|6]** *password*]

Syntax Description

username	Username of the local Cisco IP phone user. Default: Admin.
password	Enables password for the Cisco IP phone user.
password	Password string.

Command Default

Authentication credential is not created.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption, based on Unified CME password policy.

Usage Guidelines

Creates an authentication credential for SIP IP phone registration. This command is required if authentication is enabled with the **authenticate command.**

You must configure the voice register pool before configuring an authentication credential.

All lines in a phone share the same credential.

This configuration also permits the phone user to log in to Cisco Unified CME to view and change attributes associated only with the user's SIP IP phone.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]**. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Note This command is not for SIP proxy registration.

Examples

The following example shows how to set the username and password:

```
Router(config)# voice register pool 1
Router(config-register-pool)# username smith password 0 9golf
```

Related Commands

	Description
authenticate (voice register global)	Enables authentication for registration requests in which the MAC address cannot be identified by using other methods

utf8

To define Unicode UTF-8 support for a phone type, use the **utf8** command in ephone-type configuration mode. To reset to the default value, use the **no** form of this command.

utf8
no utf8

Syntax Description

This command has no arguments or keywords.

Command Default

Phone type supports Unicode UTF-8.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies whether Unicode UTF-8 is supported by the type of phone that is being added with the phone-type template.

Examples

The following example shows that UTF-8 support is set to disabled for the Nokia E61 when creating the ephone-type template:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# no utf8
```

Related Commands

Command	Description
device-id	Specifies the device ID for a phone type.
type	Assigns the phone type to an SCCP phone.

Cisco Unified CME Commands: V

vad (voice register pool)

To enable voice activity detection (VAD) on a VoIP dial peer, use the **vad** command in voice register pool configuration mode. To disable VAD, use the **no** form of this command.

vad
no vad

Syntax Description

This command has no arguments or keywords.

Command Default

VAD is enabled.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

VAD detects periods of silence in the voice signal and temporarily discontinues transmission of the signal during these periods to save bandwidth. Because VAD is enabled by default, there is no comfort noise during periods of silence. As a result, the call may seem to be disconnected and you may prefer to set **no vad** on the SIP phone pool.

Examples

The following example shows how to disable VAD for pool 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# no vad
```

vad (voice register template)

To enable voice activity detection (VAD) on SIP phones, use the **vad** command in voice register template configuration mode. To return to the default, use the **no** form of this command.

vad
no vad

Syntax Description

This command has no arguments or keywords.

Command Default

VAD is disabled.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

VAD detects periods of silence in the voice signal and temporarily discontinues transmission of the signal during these periods to save bandwidth. To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following example shows how to enable VAD:

```
Router(config)# voice register template 1
Router(config-register-temp)# vad
```

Related Commands

	Description
template (voice register pool)	Applies a template to a SIP phone.

vca

To specify the audio file used for the vacant code announcement, use the **vca** command in voice MLPP configuration mode. To disable use of this audio file, use the **no** form of this command.

vca *audio-url* **voice-class** **cause-code** *tag*
no **vca**

Syntax Description

audio-url	Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.
tag	Number of the voice class that defines the cause codes for which the VCA is played. Range: 1 to 64.

Command Default

No announcement is played.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that plays to callers when they dial an invalid or unassigned number.

The **mlpp indication** command must be enabled (default) for a phone to play precedence announcements.

The VCA plays for the cause codes defined with the **voice class cause-code** command.

This command is not supported by Cisco IOS help. If you type **?** , Cisco IOS help does not display a list of valid entries.

Examples

The following example shows that the audio file played for the vacant code announcement is named vca.au and is located in flash. The announcement plays for the unassigned-number and invalid-number cause codes, which are defined in the matching cause-code voice class.

```
voice class cause-code 1
 unassigned-number
 invalid-number
!
!
voice mlpp
 vca flash:vca.au voice-class cause-code 1
```

Related Commands

Command	Description
bnea	Specifies the audio file used for the busy station not equipped for preemption announcement.

Command	Description
bpa	Specifies the audio file used for the blocked precedence announcement.
ica	Specifies the audio file used for the isolated code announcement.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
voice class cause-code	Creates a voice class for defining a set of cause codes.

video

To enable video capability on Cisco Unified IP Phones 9951 and 9971, use the **video** command in voice register global, voice register template, and voice register pool configuration modes. To disable video capabilities on Cisco Unified IP Phones 9951 and 9971, use the **no** form of this command.

video
no video

Syntax Description

This command has no arguments or keywords.

Command Default

Video capability is disabled on Cisco Unified IP Phones 9951 and 9971.

Command Modes

Voice register global
Voice register template
Voice register pool

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.

Usage Guidelines

Use this command to enable video capability on Cisco Unified IP Phones 9951 and 9971. Video is supported on Cisco Unified IP phone 8961 through CUVA. You need to create profile and apply-config or restart to the phone to enable the video capability on phones.

Examples

The following example shows video command configured in voice register global:

```
Router#show run
!
!
!
voice service voip
 allow-connections sip to sip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
!
voice register global
 mode cme
 bandwidth video tias-modifier 244 negotiate end-to-end
 max-pool 10
 video
!
voice register template  10
!
!
```

The following example shows video command configured under voice register pool 5, you can also configure the video command under voice register template:

```
Router#show run
!
!
```

```
voice service voip
 allow-connections sip to sip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
!
!
voice register global
 mode cme
 bandwidth video tias-modifier 244 negotiate end-to-end
 max-pool 10
!
voice register pool  1
 id mac 1111.1111.1111
!
voice register pool  4
!
voice register pool  5
 logout-profile 58
 id mac 0009.A3D4.1234
 video
!
```

Related Commands

Command	Description
apply-config	Allows to dynamically apply the phone configuration on Cisco Unified SIP IP phones 8961, 9951, and 9971,
bandwidth video tias-modifier	Allows to set the maximum video bandwidth bytes per second (BPS) for SIP IP phones

video (ephone)

To enable video capabilities for an SCCP phone in Cisco Unified CME, use the **video** command in ephone configuration mode. To reset to default, use the **no** form of this command.

video
no video

Syntax Description

This command has no arguments or keywords.

Command Default

Video capabilities are disabled.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command enables video capabilities in the ephone configuration for a particular phone.

Video capabilities for SCCP phones in Cisco Unified CME must be enabled globally as well as for individual phones. You must enable video for all video-capable SCCP phones associated with a Cisco Unified CME router by configuring the videoCapability parameter of the **service phone** command.

Video parameters, such as maximum bit rate, are set at a system-level in video configuration mode.

Examples

The following example shows the ephone portion from the **show running-configuration** command:

```
router# show running-configuration
.
.
.
ephone  6
 video
 mac-address 000F.F7DE.CAA5
 type 7960
 button  1:6
```

Related Commands

service phone	Modifies the vendorConfig parameters in phone configuration files.
video (telephony-service)	Enters video configuration mode for modifying video parameters in Cisco Unified CME.

video (telephony-service)

To enter video configuration mode for setting video parameters for all video-capable phones in Cisco Unified CME, use the **video** command in telephony-service configuration mode. To reset global video parameters, use the **no** form of this command.

video
no video

Syntax Description

This command has no arguments or keywords.

Command Default

Defaults for global video parameters are configured.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command enters video configuration mode for setting video parameters for all video-capable Cisco Unified IP phones associated with a Cisco Unified CME router.

Examples

The following example shows how to enter video configuration mode for a Cisco Unified CME router. You must enter video configuration mode to set video parameters, such as maximum bit rate.

```
Router(config)#
telephony-service
Router(config-telephony)# video
Router(config-tele-video)# maximum bit-rate 256
```

Related Commands

	Description
maximum bit-rate	Sets the maximum video bandwidth for phones in Cisco unified CME.
show call active video	Displays call information for SCCP video calls in progress.
show call history video	Displays call history information for SCCP video calls.

video screening (voice service sip)

To enable transcoding and transsizing between two call legs when configuring SIP, use the **video screening** command in sip configuration mode. To disable transcoding and transsizing, use **no** form of this command.

video screening
no video screening

Syntax Description

This command has no arguments or keywords.

Command Default

Video screening is disabled

Command Modes

Sip

Command History

Release	Modification
15.1(4)M	The command was introduced.

Usage Guidelines

Use this command to enable conversion of video streams if there is a mismatch between two call legs.

Examples

The following example enters the voice-card configuration mode and enables video screening:

```
Router(config)# voice service voip
Router(config-voicecard)# sip
Router((conf-serv-sip)# video screening
```

Related Commands

Command	Description
codec profile	Defines the video capabilities needed for video endpoints.
video codec	Assigns a video codec to a VoIP dial peer.

video-bitrate (ephone)

To specify the maximum IP phone video bandwidth in Cisco Unified CME, use the **video-bitrate** command in the ephone mode. To restore the default video bitrate or suse the **no** form of this command.

video-bitrate *value*
no video-bitrate

Syntax Description

value	Video bandwidth in kb/s. Range is from 64 to 102400 kbps.

Command Default

Bit rate defaults to the maximum bit-rate configured under video configuration.

Command History

Release	Modification
15.1(4)M	This command was introduced.

Usage Guidelines

Use this command to modify the value of the maximum video bandwidth for video-capable phones that support SIP, SCCP, and H.323.

Examples

The following example sets a bit-rate of 512 kb/s.

```
Router(config)# ephone
Router(config-ephone)# video-bitrate 512
```

vm-device-id (ephone)

To define a voice-messaging identification string, use the **vm-device-id** command in ephone configuration mode. To disable this feature, use the **no** form of this command.

vm-device-id *id-string*
no vm-device-id

Syntax Description

id-string	Voice-messaging device port identification (ID) string; for example, CiscoUM-VI1 for the first port and CiscoUM-VI2 for the second port. Note that the first two characters after the hyphen must be the uppercase letters V and I.

Command Default

No voice-mail identification string is defined.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

Use this command to define a voice-messaging device ID string. A voice-messaging port registers with a device ID instead of a MAC address. To distinguish among different voice-messaging ports, the value of the voice-messaging device ID is used. The voice-messaging device ID is configured to a Cisco IP phone port, which maps to a corresponding voice-messaging port.

Examples

The following example shows how to set the voice-messaging device ID to CiscoUM-VI1:

```
Router(config) ephone 1
Router(config-ephone) vm-device-id CiscoUM-VI1
```

Related Commands

	Description
voicemail (telephony-service)	Configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.

vm-integration

To enter voice-mail integration configuration mode and enable voice-mail integration with dual tone multifrequency (DTMF) and analog voice-mail systems, use the **vm-integration** command in global configuration mode. To disable voice-mail integration, use the **no** form of this command.

vm-integration
no vm-integration

Syntax Description

This command has no arguments or keywords.

Command Default

DTMF integration with voice-mail system is disabled.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco SRST 2.1	This command was introduced for Cisco Survivable Remote Site Telephony (SRST).
12.2(2)XT	Cisco ITS 2.0	This command was introduced Cisco ITS.
12.2(8)T	Cisco ITS 2.0 Cisco SRST 2.1	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

The **vm-integration** command is used to enter voice-mail integration configuration mode to enable in-band DTMF integration with a voice-mail system.

Examples

The following example shows how to enter the voice-mail integration configuration mode:

```
Router(config) vm-integration
Router(config-vm-integration) pattern direct 2 CGN *
```

Related Commands

	Description
pattern direct (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when a user presses the Messages button on a phone.
pattern ext-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension reaches a busy extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.

	Description
pattern trunk-to-ext no-answer (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.

voice class mlpp

To create a voice class for the Multilevel Precedence and Preemption (MLPP) service, use the **voice class mlpp** command in global configuration mode. To remove the voice class, use the **no** form of this command.

voice class mlpp *tag*
no voice class mlpp *tag*

Syntax Description

tag	Unique number to identify the voice class. Range: 1 to 10000.

Command Default

No voice class is configured for MLPP.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command creates the voice class for MLPP attributes. Use the **voice-class mlpp** (dial peer) command to assign the voice class to a dial peer.

Examples

The following example shows the domain name set to DSN in the MLPP voice class:

```
Router(config)# voice class mlpp
Router(config-class)# service-domain dsn
```

Related Commands

Command	Description
service-domain (voice class)	Sets the service domain name in the MLPP voice class.
voice-class mlpp (dial peer)	Assigns an MLPP voice class to a POTS or VoIP dial peer.

voice emergency response location

To create a tag for identifying an emergency response location (ERL) for E911 services, use the **voice emergency response location** command in global configuration mode. To remove the ERL tag, use the **no** form of this command.

voice emergency response location *tag*
no voice emergency response location *tag*

Syntax Description

tag	Unique number that identifies this ERL tag.

Command Default

No ERL tag is created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added for Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to create an ERL that identifies an area where emergency teams can quickly locate a 911 caller. The ERL definition optionally includes which ELINs are associated with the ERL and which IP phones are located in the ERL. You can define two or fewer unique IP subnets and two or fewer ELINs. If you define one ELIN, this ELIN is always used for phones calling from this ERL. If you define two ELINs, the system alternates between using each ELIN. If you define zero ELINs and phones use this ERL, the outbound calls do not have their calling numbers translated. The PSAP sees the original calling numbers for these 911 calls. You can optionally add the civic address using the **address** command and an address description using the **name** command.

Examples

In the following example, all IP phones with the IP address of 10.X.X.X or 192.168.X.X are automatically associated with this ERL. If one of the phones dials 911, its extension is replaced with 408 555-0100 before it goes to the PSAP. The PSAP will see that the caller’s number is 408 555-0100. The civic address, 410 Main St, Tooly, CA, and a descriptive identifier, Bldg 3 are included.

```
voice emergency response location 1
 elin 1 4085550100
 subnet 1 10.0.0.0 255.0.0.0
 subnet 2 192.168.0.0 255.255.0.0
 address 1,408,5550100,410,Main St.,Tooly,CA
 name Bldg 3
```

Related Commands

Command	Description
address	Specifies a comma separated text entry (up to 250 characters) of an ERL's civic address.
elin	Specifies a PSTN number that will replace the caller's extension.
name	Specifies a string (up to 32-characters) used internally to identify or describe the emergency response location.
subnet	Defines which IP phones are part of this ERL.

voice emergency response settings

To define 911 call behavior settings, use the **voice emergency response settings** command in global configuration mode. To remove the settings, use the **no** form of this command.

voice emergency response settings
no voice emergency response settings

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to enable definition of the following 911 call behavior settings:

- **elin**: Default ELIN to use if a 911 caller's IP phone's address does not match the subnet of any location in any zone.
- **expiry**: Number of minutes a 911 call is associated to an ELIN in the case of a callback from the 911 operator.
- **callback**: Default number to contact if a 911 callback cannot find the last 911 caller.
- **logging**: Syslog informational message that is printed to the console each time an emergency call is made. This feature is enabled by default, however you can disable this feature by entering the **no** form of this command.

Examples

In the following example, if the 911 caller's IP phone address does not match any of the voice emergency response locations, the ELIN defined in the **voice emergency response settings** configuration (4085550101) is used. After the 911 call is placed to the PSAP, the PSAP has 120 minutes (2 hours) to call back 408 555-0101 to reach the 911 caller. If during a callback, the last caller's extension number cannot be found, the call is routed to extension 7500. The outbound 911 calls do not cause a syslog message to the logging facility (for example, to the local buffer, console, or remote host).

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
no logging
```

Related Commands

Command	Description
callback	Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.
elin	E.164 number used as the default ELIN if no matching ERL to the 911 caller's IP phone address is found.
expiry	Number of minutes a 911 call is associated to an ELIN in the case of a callback from the 911 operator.
logging	Syslog informational message printed to the console every time an emergency call is made.

voice emergency response zone

To create an emergency response zone, use the **voice emergency response zone** command in global configuration mode. To remove the created voice emergency response zone, use the **no** form of this command.

voice emergency response zone *tag*
no voice emergency response zone *tag*

Syntax Description

tag	Identifier (1-100) for the voice emergency response zone.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to create voice emergency response zones that allow routing of 911 calls to different PSAPs.

Examples

The following example shows an assignment of ERLs to a voice emergency response zone. The calls have an ELIN from ERLs 8, 9, and 10. The locations for ERLs in zone 10 are searched in the order each CLI is entered for a phone address match because no priority order is assigned.

```
voice emergency response zone 10
location 8
location 9
location 10
```

Related Commands

Command	Description
location	Identifies locations within an emergency response zone and optionally assigns a priority order to the location.

voice hunt-group

To create a hunt group for phones in a Cisco Unified Communications Manager Express (Cisco Unified CME) or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) system, use the **voice hunt-group** command in global configuration mode. To delete a hunt group, use the **no** form of this command.

voice hunt-group *hunt-tag* {**longest-idle** | **parallel** | **peer** | **sequential**}
no voice hunt-group *hunt-tag*

Syntax Description

hunt-tag	Unique sequence number that identifies the hunt group. Range is 1 to 100.
longest-idle	Allows an incoming call to go first to the number that has been idle the longest for the number of hops specified when the hunt group was defined. The longest-idle time is determined from the last time that a phone registered, reregistered, or went on-hook.
parallel	Allows an incoming call to simultaneously ring all the numbers in the hunt group member list.
peer	Allows a round-robin selection of the first extension to ring. Ringing proceeds in a circular manner from left to right. The round-robin selection starts with the number left of the number that answered when the hunt-group was last called.
sequential	Allows an incoming call to ring all the numbers in the left-to-right order in which they were listed when the hunt group was defined.

Command Default

By default, voice hunt group is not created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was modified to add support for Cisco Unified SCCP IP phones.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.2(4)M	Cisco Unified SIP SRST 9.1	This command was introduced in Cisco Unified SIP SRST 9.1.
15.3(4)M	Cisco Unified CME 10.5	This command was modified to include support for wildcards which is indicated by "*" . symbol.

Usage Guidelines

The **voice hunt-group** command enters voice hunt-group configuration mode to define a hunt group. A hunt group is a list of phone numbers that take turns receiving incoming calls to a specific number (pilot number), which is defined with the **pilot** command. The specific extensions included in the hunt group and the order and maximum number of extensions allowed in the list are defined with the **list** command.

If a number in the list is busy or does not answer, the call is redirected to the next number in the list. The last number tried is the final number, which is defined with the **final** command. If the number of times that a call is redirected to a new number exceeds 5, you must use the **max-redirect** command to increase the allowable number of redirects in the Cisco Unified CME or Cisco Unified SIP SRST system.

To configure a new hunt group, you must specify the **longest-idle**, **parallel**, **peer**, or **sequential** keyword. To change an existing hunt group configuration, the keyword is not required. To change the type of hunt group, for instance from peer to sequential or sequential to peer, you must remove the existing hunt group first by using the **no** form of this command and then re-create it.

The **parallel** keyword creates a dial peer to allow an incoming call to ring multiple phones simultaneously. The use of parallel hunt groups is also referred to as application-level forking because it enables the forking of a call to multiple destinations. A pilot dial peer cannot be used as a voice hunt group and a hunt group at the same time.

While ephone hunt groups only support Cisco Unified SCCP IP phones, a voice hunt group supports either a Cisco Unified SCCP IP phone or a Cisco Unified SIP IP phone.

With the voice hunt group feature preconfigured in the Cisco Unified SIP SRST router, voice hunt groups continue to be supported after phones fallback from a Cisco Unified Communications Manager (Cisco Unified CM) to a Cisco Unified SIP SRST router.

Examples

The following example shows how to define longest-idle hunt group 1 with pilot number 7501, final number 8000, and nine numbers in the list. After a call is redirected six times (makes 6 hops), it is redirected to the final number 8000.

```
Router(config)# voice hunt-group 1 longest-idle
Router(config-voice-hunt-group)# pilot 7501
Router(config-voice-hunt-group)# list 7001, 7002, 7023, 7028, 7045, 7062, 7067, 7072, 7079
 Router(config-voice-hunt-group)# final 8000

Router(config-voice-hunt-group)# hops 6
Router(config-voice-hunt-group)# timeout 20

Router(config-voice-hunt-group)# exit
```

The following example shows how to define peer hunt group number 2. Callers dial the pilot number 5610 to reach the hunt group. The first extension to ring the first time that this hunt group is called is 5601. If 5601 does not answer, the hunt proceeds from left to right, beginning with the extension directly to the right. If none of those extensions answer, the call is forwarded to extension 6000, which is the number for the voice-mail service.

The second time someone calls the hunt group, the first extension to ring is 5602 if 5601 was answered during the previous call.

```
Router(config)# voice hunt-group 2 peer
Router(config-voice-hunt-group)# pilot 5610
Router(config-voice-hunt-group)# list 5601, 5602, 5617, 5633

Router(config-voice-hunt-group)# final 6000
Router(config-voice-hunt-group)# timeout 30

Router(config-voice-hunt-group)# exit
```

The following example shows how to define sequential hunt group number 3. When callers dial extension 5601, the first phone to ring is 5001, then 5002, 5017, and 5028. If none of those extensions answer, the call is forwarded to extension 6000, which is the number for the voice-mail service.

```
Router(config)# voice hunt-group 3 sequential
Router(config-voice-hunt-group)# pilot 5601
Router(config-voice-hunt-group)# list 5001, 5002, 5017, 5028

Router(config-voice-hunt-group)# final 6000
Router(config-voice-hunt-group)# timeout 30

Router(config-voice-hunt-group)# exit
```

The following example shows how to define a parallel hunt group. When callers dial extension 1000, extensions 1001, 1002, and so forth ring simultaneously. The first extension to answer is connected. All other call legs are disconnected. If none of the extensions answer, the call is forwarded to extension 2000, which is the number for the voice-mail service.

```
Router(config)# voice hunt-group 4 parallel
Router(config-voice-hunt-group)# pilot 1000
Router(config-voice-hunt-group)# list 1001, 1002, 1003, 1004
Router(config-voice-hunt-group)# final 2000

Router(config-voice-hunt-group)# timeout 20

Router(config-voice-hunt-group)# exit
```

The following example shows the support for wildcard slots in voice hunt-groups.

```
Router(config)#Voice hunt-group 1 parallel
Router(config-voice-hunt-group)#pilot number 100
Router(config-voice-hunt-group)#List 1001, 1002, 1002, *, *
Router(config-voice-hunt-group)# exit
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next phone number in a peer voice hunt-group list before proceeding to the final phone number.
list (voice hunt-group)	Defines the phone numbers that participate in a voice hunt group.
max-redirect	Changes the number of times that a call can be redirected by call forwarding or transfer within a Cisco Unified CME system.
pilot (voice hunt-group)	Defines the phone number that callers dial to reach a voice hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last phone number in the hunt group.

voice-hunt-groups login

To enable a voice register dn or ephone dn to join or unjoin voice hunt-groups dynamically, use the **voice-hunt-groups login** command in voice register dn configuration mode. To disable this capability, use the **no** form of this command.

voice-hunt-groups login
no voice-hunt-groups login

Syntax Description

This command has no arguments or keywords.

Command Default

A voice register dn or ephone dn is not allowed to dynamically join and unjoin voice hunt groups.

Command Modes

voice register dn configuration (config-voice-register-dn)

ephone dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified CME 10.5	This command was introduced.

Usage Guidelines

Use the **show voice hunt-groups** command to display current hunt group members, including those who joined the group dynamically.

Examples

The following example creates five voice register dns and a hunt group that includes the first two voice register dn and two wildcard slots. The last three voice register dns are enabled for voice hunt group dynamic membership. Each of them can join and unjoin the hunt group whenever one of the slots is available.

```
voice register dn 22
 number 4566
voice register dn 23
 number 4567
voice register dn 24
 number 4568
 voice-hunt-groups login
voice register dn 25
 number 4569
 voice-hunt-groups login
voice register dn 26
 number 4570
 voice-hunt-groups login
voice-hunt-groups 1 peer
 list 4566,4567,*,*
 final 7777
```

The following example creates three ephone dns and a hunt group that includes the first two ephone dn and two wildcard slots. The last one ephone dn is enabled for voice hunt group dynamic membership. Each of them can join and unjoin the hunt group whenever one of the slots is available

```
ephone-dn 22
 number 4566
ephone-dn 23
 number 4567
ephone-dn 24
 number 4568
 voice-hunt-groups login
voice-hunt-groups 1 peer
 list 4566,4567,*,*
 final 7777
```

Related Commands

Command	Description
show voice hunt-groups	Displays voice-hunt group configuration, current status, and statistics.

voice lpcor call-block cause

To define the cause code that is used when a call is blocked because LPCOR validation fails, use the **voice lpcor call-block cause** command in global configuration mode. To reset to the default, use the **no** form of this command.

voice lpcor call-block cause *cause-code*
no voice lpcor call-block cause

Syntax Description

cause-code	Number of the cause code to generate when a call is blocked by the LPCOR validation process. Range: 1 to 180.

Command Default

Default cause code is 63 (serv/opt-unavail-unspecified).

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

The following table lists the available cause codes.

Table 76: Cause Codes for Calls Blocked by LPCOR Validation

Message	Description	Code Number
access-info-discard	access info discarded (43)	43
b-cap-not-implemented	bearer capability not implemented (65)	65
b-cap-restrict	restricted digital info bc only (70)	70
b-cap-unauthorized	bearer capability not authorized (57)	57
b-cap-unavail	bearer capability not available (58)	58
call-awarded	call awarded (7)	7
call-cid-in-use	call exists call id in use (83)	83
call-clear	call cleared (86)	86
call-reject	call rejected (21)	21
cell-rate-unavail	cell rate not available (37)	37
channel-unacceptable	channel unacceptable (6)	6
chantype-not-implement	chan type not implemented (66)	66

Message	Description	Code Number
cid-in-use	call id in use (84)	84
codec-incompatible	codec incompatible (171)	171
cug-incalls-bar	cug incoming calls barred (55)	55
cug-outcalls-bar	cug outgoing calls barred (53)	53
dest-incompatible	incompatible destination (88)	88
dest-out-of-order	destination out of order (27)	27
dest-unroutable	no route to destination (3)	3
dsp-error	dsp error (172)	172
dtl-trans-not-node-id	dtl transit not my node id (160)	160
facility-not-implemented	facility not implemented (69)	69
facility-not-subscribed	facility not subcribed (50)	50
facility-reject	facility rejected (29)	29
glare	glare (15)	15
glaring-switch-pri	glaring switch PRI (180)	180
htspm-oos	HTSPM out of service (129)	129
ie-missing	mandatory ie missing (96)	96
ie-not-implemented	ie not implemented (99)	99
info-class-inconsistent	inconsistency in info and class (62)	62
interworking	interworking (127)	127
invalid-call-ref	invalid call ref value (81)	81
invalid-ie	invalid ie contents (100)	100
invalid-msg	invalid message (95)	95
invalid-number	invalid number (28)	28
invalid-transit-net	invalid transit network (91)	91
misdialled-trunk-prefix	misdialled trunk prefix (5)	5
msg-incomp-call-state	message in incomp call state (101)	101
msg-not-implemented	message type not implemented (97)	97
msgtype-incompatible	message type not compatible (98)	98

Message	Description	Code Number
net-out-of-order	network out of order (38)	38
next-node-unreachable	next node unreachable (128)	128
no-answer	no user answer (19)	19
no-call-suspend	no call suspended (85)	85
no-channel	channel does not exist (82)	82
no-circuit	no circuit (34)	34
no-cug	non existent cug (90)	90
no-dsp-channel	no dsp channel (170)	170
no-req-circuit	no requested circuit (44)	44
no-resource	no resource (47)	47
no-response	no user response (18)	18
no-voice-resources	no voice resources available (126)	126
non-select-user-clear	non selected user clearing (26)	26
normal-call-clear	normal call clearing (16)	16
normal-unspecified	normal unspecified (31)	31
not-in-cug	user not in cug (87)	87
number-changeed	number changed (22)	22
param-not-implemented	non implemented param passed on (103)	103
perm-frame-mode-oos	perm frame mode out of service (39)	39
perm-frame-mode-oper	perm frame mode operational (40)	40
precedence-call-block	precedence call blocked (46)	46
preempt	preemption (8)	8
preempt-reserved	preemption reserved (9)	9
protocol-error	protocol error (111)	111
qos-unavail	qos unavailable (49)	49
rec-timer-exp	recovery on timer expiry (102)	102
redirect-to-new-destination	redirect to new destination (23)	23
req-vpci-vci-unavail	requested vpci vci not available (35)	35

Message	Description	Code Number
send-infotone	send info tone (4)	4
serv-not-implemented	service not implemented (79)	79
serv/opt-unavail-unspecified	service or option not available unspecified (63)	63
stat-enquiry-resp	response to status enquiry (30)	30
subscriber-absent	subscriber absent (20)	20
switch-congestion	switch congestion (42)	42
temp-fail	temporary failure (41)	41
transit-net-unroutable	no route to transit network (2)	2
unassigned-number	unassigned number (1)	1
unknown-param-msg-discard	unrecognized param msg discarded (110)	110
unsupported-aal-parms	aal parms not supported (93)	93
user-busy	user busy (17)	17
vpci-vci-assign-fail	vpci vci assignment failure (36)	36
vpci-vci-unavail	no vpci vci available (45)	45

Examples

The following example shows the cause code set to 79:

```
Router(config)# voice lpcor call-block cause 79
```

Related Commands

Command	Description
voice lpcor policy	Creates a LPCOR policy for a resource group.

voice lpcor custom

To define the logical partitioning class of restriction (LPCOR) resource groups on the Cisco Unified CME router, use the **voice lpcor custom** command in global configuration mode. To remove the custom resource list, use the **no** form of this command.

voice lpcor custom
no voice lpcor custom

Syntax Description

This command has no arguments or keywords.

Command Default

Custom LPCOR resource list is not defined.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command enters LPCOR custom configuration mode where you define the name of each of your resource groups using the **index** command. Only one custom resource list is allowed on a Cisco Unified CME router. After you add a resource group to this list, you must then create a LPCOR policy for each resource group that requires call restrictions.

Examples

The following example shows a LPCOR configuration with six resource groups:

```
voice lpcor custom
 group 1 sccp_phone_local
 group 2 sip_phone_local
 group 3 analog_phone_local
 group 4 sip_remote
 group 5 sccp_remote
 group 6 isdn_local
```

Related Commands

Command	Description
group (lpcor custom)	Adds a LPCOR resource group to the custom resource list.
voice lpcor enable	Enables LPCOR functionality on the Cisco Unified CME router.
voice lpcor policy	Creates a LPCOR policy for a resource group.

voice lpcor enable

To enable logical partitioning class of restriction (LPCOR) functionality on the Cisco Unified CME router, use the **voice lpcor enable** command in global configuration mode. To reset to the default, use the **no** form of this command.

voice lpcor enable
no voice lpcor enable

Syntax Description

This command has no arguments or keywords.

Command Default

LPCOR capability is disabled.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

After using this command, use the **voice lpcor custom** command to create a list of your LPCOR resource groups.

Examples

The following example shows a configuration with LPCOR enabled and a custom resource list :

```
voice lpcor enable
voice lpcor custom
 group 1 local_sccp_phone_1
 group 2 local_sip_phone_1
 group 3 local_analog_phone_1
 group 4 local_sccp_phone_2
!
voice lpcor policy local_sccp_phone_1
 accept local_sip_phone_1
 accept local_analog_phone_1
 accept local_sccp_phone_2
```

Related Commands

Command	Description
voice lpcor custom	Defines the LPCOR resource groups on the Cisco Unified CME router.
voice lpcor policy	Creates a LPCOR policy for a resource group.

voice lpcor ip-phone mobility

To set the default LPCOR policy for mobility-type phones, use the **voice lpcor ip-phone mobility** command in global configuration mode. To reset to the default, use the **no** form of this command.

voice lpcor ip-phone mobility {**incoming** | **outgoing**} *lpcor-group*
no voice lpcor ip-phone mobility {**incoming** | **outgoing**}

Syntax Description

incoming	Sets default LPCOR policy for incoming calls.
outgoing	Sets default LPCOR policy for outgoing calls.
lpcor-group	Name of the LPCOR resource group.

Command Default

Default LPCOR policy is not defined for mobility-type phones.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command defines the default LPCOR policy for a mobility-type phone if the LPCOR policy cannot be provisioned using the LPCOR IP-phone subnet table.

Examples

The following example shows that the default LPCOR policy for mobility-type phones is set to remote_group1. Any mobility-type phones with a shared IP address from DHCP pool1 are considered local IP phones and are associated with the local_group1 LPCOR policy. Other mobility-type phones without a shared IP address are considered remote IP phones and are associated with the remote_group1 default LPCOR policy.

```
voice lpcor ip-phone subnet incoming
 index 1 local_group1 dhcp-pool pool1
!
voice lpcor ip-phone subnet outgoing
 index 1 local_group1 dhcp-pool pool1
!
voice lpcor ip-phone mobility incoming remote_group1
voice lpcor ip-phone mobility outgoing remote_group1
```

Related Commands

Command	Description
voice lpcor ip-phone subnet	Creates a LPCOR IP-phone subnet table for calls to or from a mobility-type phone.

voice lpcor ip-phone subnet

To create a logical partitioning class of restriction (LPCOR) IP-phone subnet table for calls to or from a mobility-type phone, use the **voice lpcor ip-phone subnet** command in global configuration mode. To reset to the default, use the **no** form of this command.

voice lpcor ip-phone subnet {**incoming** | **outgoing**}
no voice lpcor ip-phone subnet {**incoming** | **outgoing**}

Syntax Description

incoming	Creates IP-phone subnet table for incoming calls from mobility-type phone.
outgoing	Creates IP-phone subnet table for outgoing calls from mobility-type phone.

Command Default

IP-phone subnet table is not created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command is used for mobility-type phones only, which can include Extension Mobility phones, teleworker remote phones, and Cisco IP Communicator softphones.

This command enters LPCOR IP-phone subnet configuration mode to add LPCOR groups to the incoming or outgoing IP-phone subnet tables. Two IP-phone subnet tables, one for incoming calls and one for outgoing calls, can be defined on each Cisco Unified CME router and can include up to 50 IP address or DHCP pool entries.

A LPCOR policy is dynamically associated with calls to and from a mobility-type phone based on its current IP address or DHCP pool.

Examples

The following example shows :

```
voice lpcor ip-phone subnet incoming
 index 1 local_g2 10.0.10.23 255.255.255.0 vrf vrf-group2
 index 2 remote_g2 171.19.0.0 255.255.0.0
 index 3 local_g1 dhcp-pool pool1

voice lpcor ip-phone subnet outgoing
 index 1 local_g4 10.1.10.23 255.255.255.0 vrf vrf-group2
 index 2 remote_g4 171.19.0.0 255.255.0.0
 index 3 local_g5 dhcp-pool pool1
```

Related Commands

Command	Description
index (ip-phone)	Adds a LPCOR group to the IP-phone subnet table.

Command	Description
lpcor type	Specifies the LPCOR type for an IP phone.
voice lpcor ip-phone mobility	Sets the default LPCOR policy for mobility-type phones.

voice lpcor ip-trunk subnet incoming

To create a logical partitioning class of restriction (LPCOR) IP-trunk subnet table for incoming calls from a VoIP trunk, use the **voice lpcor ip-trunk subnet incoming** command in global configuration mode. To reset to the default, use the **no** form of this command.

voice lpcor ip-trunk subnet incoming
no voice lpcor ip-trunk subnet incoming

Syntax Description

This command has no arguments or keywords.

Command Default

IP-trunk subnet table is not created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command enters LPCOR IP-trunk subnet configuration mode to add LPCOR groups to the IP-trunk subnet table. One IP-trunk subnet table, containing up to 50 index entries, can be defined on each Cisco Unified CME router for incoming calls from H.323 or SIP trunks.

Incoming VoIP trunk calls are associated with a LPCOR policy by matching the IP address or hostname in the IP-trunk subnet table first. If the IP address or hostname is not found in the table, the LPCOR policy specified with the **lpcor incoming** command in voice service configuration mode is applied.

Examples

The following example shows three resource groups are included in the IP-trunk subnet table:

```
voice lpcor ip-trunk subnet incoming
 index 1 h323_group1 172.19.33.0 255.255.255.0
 index 2 sip_group1 172.19.22.0 255.255.255.0
 index 3 sip_group2 hostname sipexample
```

Related Commands

Command	Description
index (lpcor ip-trunk)	Adds a LPCOR resource group to the IP trunk subnet table.
lpcor incoming	Associates a LPCOR resource-group policy with an incoming call.
voice lpcor ip-phone subnet	Creates a LPCOR IP-phone subnet table for calls to or from a mobility-type phone.

voice lpcor policy

To create a logical partitioning class of restriction (LPCOR) policy for a resource group, use the **voice lpcor policy** command in global configuration mode. To reset to the default, use the **no** form of this command.

voice lpcor policy *lpcor-group*
no voice lpcor policy *lpcor-group*

Syntax Description

lpcor-group	Name of the LPCOR resource group.

Command Default

LPCOR policy is not defined.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

You can define one policy for each LPCOR resource group. The policy defines the other resource groups from which this resource group can accept calls. You must first name the policy by including it in the custom resource list using the **voice lpcor custom** command.

If you do not explicitly include any resource groups in the policy by using the **accept** command, that policy blocks all incoming calls that are associated with any LPCOR policy other than its own.

If a LPCOR policy is not defined for a target destination, the target can accept incoming calls from any resource group.

Examples

The following examples show a LPCOR configuration with four resource groups:

```
voice lpcor custom
 index 1 siptrunk
 index 2 h323trunk
 index 3 pstn
 index 4 voicemail
!
```

The LPCOR policy for h323trunk accepts calls from the voicemail group and rejects calls from the siptrunk and pstn groups:

```
voice lpcor policy h323trunk
 accept voicemail
!
```

The LPCOR policy for pstn blocks calls from the siptrunk, h323trunk, and voicemail groups:

```
voice lpcor policy pstn
!
```

The LPCOR policy for voicemail accepts calls from the siptrunk, h323trunk, and pstn groups:

```
voice lpcor policy voicemail
 accept siptrunk
 accept h323trunk
 accept pstn
```

The siptrunk group does not have a LPCOR policy defined so it can accept calls from any of the other resource groups.

Related Commands

Command	Description
accept	Allows a LPCOR resource group to accept incoming calls from another resource group.
show voice lpcor policy	Displays the LPCOR policy for the specified resource group.
voice lpcor custom	Defines the LPCOR resource groups on the Cisco Unified CME router.

voice mlpp

To enter MLPP configuration mode to enable MLPP service, use the voice service command in global configuration mode. To disable MLPP service, use the **no** form of this command.

voice mlpp
no voice mlpp

Syntax Description

This command has no keywords or arguments.

Command Default

No default behavior or values.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Products	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Voice-mlpp configuration mode is used for the gateway globally.

Examples

The following example shows how to enter voice-mlpp configuration mode:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# access-digit
```

Related Commands

Command	Description
access-digit	Defines the access digit that phone users dial to request a precedence call.
mlpp preemption	Enables calls on an SCCP phone or analog FXS port to be preempted.
preemption trunkgroup	Enables preemption capabilities on a trunk group.

voice moh-group

To enter voice-moh-group configuration mode and set up music on hold (MOH) group parameters, use the **voice moh-group** command in global configuration mode. To remove the music on hold (MOH) group parameters from the configuration for SCCP IP phones, use the **no** form of this command.

voice moh-group moh-group tag
no voice moh-group tag

Syntax Description

tag	Specifies a moh-group number tag (1-5) to be used for music on hold group parameters.

Command Default

No voice-moh-group is enabled.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command enters the voice-moh-group configuration mode for configuring music on hold (MOH) group parameters for SCCP IP phones in Cisco Unified CME or in Cisco Unified SRST.

Examples

The following example shows how to enter voice-moh-group configuration mode for configuring a moh group in Cisco Unified CME. This example also includes the command to configure a music on hold (MOH) flash file for this voice-moh- group.

```
Router(config)# voice-moh-group 1
Router(config-voice-moh-group)#moh minuet.wav
```

Related Commands

moh	Enables music on hold from a flash audio feed.
multicast moh	Enables multicast of the music-on-hold audio stream.
extension-range	Defines extension range for a clients calling a voice-moh-group.

voice register dialplan

To enter voice register dialplan configuration mode to define a dial plan for SIP phones, use the **voice register dialplan** command in global configuration mode. To remove the dialplan, use the **no** form of this command.

voice register dialplan *dialplan-tag*
no voice register dialplan *dialplan-tag*

Syntax Description

dialplan-tag	Number that identifies the dial plan. Range: 1 to 24.

Command Default

No dial plan is defined.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

A dial plan allows a SIP phone to determine when enough digits are collected for call processing to take place. You define a dial plan using this command and then apply the dial plan to a SIP phone by using the **dialplan** command.

Dial plans allow SIP phones to perform pattern recognition as user input is collected. After a defined pattern is recognized, a SIP INVITE message is automatically sent to Cisco Unified CME and the user does not have to press the Dial key or wait for the interdigit timeout.

This command creates a dial plan file that is downloaded to the phone when the phone is reset or restarted.

Examples

The following example shows how to create dial plan 10 for a Cisco Unified IP Phone 7905:

```
Router(config)# voice register dialplan 10
Router(config-register-dialplan)# type 7905-7912
Router(config-register-dialplan)# pattern 52...
Router(config-register-dialplan)# pattern 91.......
```

Related Commands

	Description
dialplan	Assigns a dial plan to a SIP phone.
filename	Specifies a custom XML configuration file that contains the dial patterns to use for a SIP dial plan.
pattern (voice register dialplan)	Defines a dial pattern for a SIP dial plan.
show voice register dialplan	Displays all configuration information for a specific SIP dial plan.

	Description
type (voice register dialplan)	Defines a phone type for a SIP dial plan.

voice register dn

To enter voice register dn configuration mode to define an extension for a phone line, intercom line, voice-mail port, or a message-waiting indicator (MWI), use the **voice register dn** command in global configuration mode. To remove the directory number, use the **no** form of this command.

voice register dn *dn-tag*
no voice register dn *dn-tag*

Syntax Description

dn-tag	Unique sequence number that identifies a particular directory number during configuration tasks. Range is 1 to 150, or the maximum defined by the **max-dn** command.

Command Default

Directory number is not defined.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

Use this command to create directory numbers for SIP IP phones directly connected in Cisco Unified CME. In voice register dn configuration mode, you assign an extension number by using the **number** command, a name to appear in the local directory by using the **name** command, and other provisioning parameters by using various commands.

Before using this command, set the maximum number of directory numbers to appear in your system by using the **max-dn** command in voice register global configuration mode.

Note This command can also be used for Cisco SIP SRST.

Note The name or label associated with a directory number configured under **voice register dn** configuration mode cannot contain special characters such as quotes ("), angle brackets (<, >), ampersand (&), and percentage (%).

Examples

The following example shows how to enter voice register dn configuration mode for directory number 4 and forward calls to extension 8888 when extension 1001 does not answer:

```
Router(config)# voice register dn 4
Router(config-register-dn)# number 1001
Router(config-register-dn)# call-forward phone noan 8888
Router(config-register-dn)# call-forward b2bua all 5454

Router(config-register-dn)# call-forward b2bua busy 5705

Router(config-register-dn)# call-forward b2bua mbox 5550
```

```
Router(config-register-dn)# call-forward b2bua noan 5050 timeout 20

Router(config-register-dn)# after-hour exempt
```

Related Commands

	Description
max-dn (voice register global)	Sets the maximum number of SIP phone directory numbers (extensions) supported by a Cisco CME router.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.
number (voice register pool)	Configures a valid number for a SIP phone.

voice register global

To enter voice register global configuration mode in order to set global parameters for all supported Cisco SIP IP phones in a Cisco Unified CME or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment, use the **voice register global** command in global configuration mode. To automatically remove the existing DNs, pools, and global dialplan patterns, use the **no** form of this command.

voice register global
no voice register global

Syntax Description

This command has no arguments or keywords.

Command Default

There are no system-level parameters configured for SIP IP phones.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
15.0(1)XA	Cisco SIP SRST 8.0	This command was updated to display the signaling transport protocol.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	The no form of the command was modified.

Usage Guidelines

Cisco Unified CME

Use this command to set provisioning parameters for all supported SIP phones in a Cisco Unified CME system.

Cisco Unified SIP SRST

Use this command to set provisioning parameters for multiple pools; that is, all supported Cisco SIP IP phones in a SIP SRST environment.

Cisco Unified CME 8.1 enhances the no form of voice register global command. The no voice register global command clears global configuration along with pools and DN configuration and also removes the configurations for voice register template, voice register dialplan, and voice register session-server. A confirmation is sought before the cleanup is made.

In Cisco Unified SRST 8.1 and later versions, the no voice register global command removes pools and DNs along with the global configuration.

Note The name or label associated with a directory number configured under **voice register global** configuration mode cannot contain special characters such as quotes ("), angle brackets (<, >), ampersand (&), and percentage (%).

Examples

Cisco Unified CME

The following is a partial sample output from the **show voice register global** command. All of the parameters listed were set under voice register global configuration mode:

```
Router# show voice register global
CONFIG [Version=4.0(0)]
========================
Version 4.0(0)
Mode is cme
Max-pool is 48
Max-dn is 48
Source-address is 10.0.2.4 port 5060
Load 7960-40 is P0S3-07-4-07
Time-format is 12
Date-format is M/D/Y
Time-zone is 5
Hold-alert is disabled
Mwi stutter is disabled
Mwi registration for full E.164 is disabled
Dst auto adjust is enabled
 start at Apr week 1 day Sun time 02:00
 stop  at Oct week 8 day Sun time 02:00
```

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from no voice register global command:

```
Router(config)# no voice register global
This will remove all the existing DNs, Pools, Templates,
Dialplan-Patterns, Dialplans and Feature Servers on the system.
Are you sure you want to proceed? Yes/No? [no]:
```

Related Commands

Command	Description
allow connections sip to sip	Allows connections between SIP endpoints in a Cisco multiservice IP-to-IP gateway.
application (voice register global)	Selects the session-level application for all dial peers associated with SIP phones.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified system.

voice register pool

To enter voice register pool configuration mode and create a pool configuration for a SIP IP phone in Cisco Unified CME or for a set of SIP phones in Cisco Unified SIP SRST, use the **voice register pool** command in global configuration mode. To remove the pool configuration, use the **no** form of this command.

voice register pool *pool-tag*
no voice register pool *pool-tag*

Syntax Description

pool-tag	Unique number assigned to the pool. Range is 1 to 100. **Note** For Cisco Unified CME systems, the upper limit for this argument is defined by the **max-pool** command.

Command Default

There is no pool configured.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.

Usage Guidelines

Cisco Unified CME

Use this command to set phone-specific parameters for SIP phones in a Cisco Unified CME system. Before using this command, enable the **mode cme** command and set the maximum number of SIP phones supported in your system by using the **max-pool** command.

Cisco Unified SIP SRST

Use this command to enable user control on which registrations are to be accepted or rejected by a SIP SRST device. The voice register pool command mode can be used for specialized functions and to restrict registrations on the basis of MAC, IP subnet, and number range parameters.

Examples

Cisco Unified CME

The following example shows how to enter voice register pool configuration mode and forward calls to extension 9999 when extension 2001 is busy:

```
Router(config)# voice register pool 10
Router(config-register-pool)# type 7960
Router(config-register-pool)# number 1 2001
Router(config-register-pool)# call-forward busy 9999 mailbox 1234
```

Cisco Unified SIP SRST

The following partial sample output from the **show running-config** command shows that several voice register pool commands are configured within voice register pool 3:

```
voice register pool 3
 id network 10.2.161.0 mask 255.255.255.0
 number 1 95... preference 1
 cor outgoing call95 1 95011
 max registrations 5
 voice-class codec 1
```

Related Commands

	Description
max-pool (voice register global)	Sets the maximum number of SIP phones that are supported by a Cisco Unified CME system.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
number (voice register pool)	Configures a valid number for a SIP phone.
type (voice register pool)	Defines a Cisco IP phone type.

voice register pool-type

To enter voice register pool-type configuration mode and add a new Cisco Unified SIP IP phone to Cisco Unified CME, use the **voice register pool-type** command in global configuration mode. To return to the default, use the **no** form of this command.

voice register pool-type **[device-reference***supported phone-type***]** **[device-name***name***{device-type***type***}[addons***max-addons***]{num-lines***max-lines***}[transport-type {udp | tcp}][gsm-handoff][telnet][phoneload][xml-config***xml-config value***}**

novoice register pool-type **[device-reference***supported phone-type***]** **[device-name***name***{device-type***type***}[addons***max-addons***]{num-lines***max-lines***}[transport-type {udp | tcp}][gsm-handoff][telnet][phoneload][xml-config***xml-config value***}**

Syntax Description

device-reference *supported phone-type*	(Optional) Defines the nearest-supported phone from which a new Cisco Unified SIP IP phone can inherit properties without the explicit configuration of the parameters. For a list of the names, types, and corresponding properties of supported phones, see the below table.
device-name *name*	(Optional) Defines the description string for the new phone device.
device-name *type*	Defines a phone type for a new Cisco Unified SIP IP phone.
addons *max-addons*	(Optional) Defines the maximum number of add-on modules supported by the new phone device. The maximum allowed value is 3. **Note** New add-on modules for an existing phone are not supported.
num-lines *max-lines*	Defines the maximum number of lines supported by the phone. Range is 1 to ???.
transport-type {udp | tcp}	(Optional) Defines the transport protocol supported by the phone. • udp—User Datagram Protocol (UDP) is used. • tcp—Transmission Control Protocol (TCP) is used.
gsm-handoff	(Optional) Enables phone support for Global System for Mobile Communications (GSM) handoff. **Note** For Cisco IOS Release 15.3(3)M, only CiscoMobile-iOS and Jabber-Android are supported.
telnet	(Optional) Enables phone support for Telnet access. **Note** For Cisco IOS Release 15.3(3)M, only Cisco Unified 3911, 3951, 7905, 7912, 7960, and 7940 SIP IP phones support Telnet access.
phoneload	(Optional) Enables support for phone loads.

xml-config *xml-tag value*	(Optional) Defines the phone-specific XML tags to be used in the configuration file. • xml-tag—Phone-specific XML tag. • value—Value of the XML tag.

Command Default

The Cisco Unified 7965 SIP IP phone model is used as the default phone device reference.

Command Modes

Global configuration (config)

Command History

Release	Modification
15.3(3)M	This command was introduced.

Usage Guidelines

When the device-reference keyword is not configured and phone properties are not explicitly configured, the Cisco Unified 7965 SIP IP phone model is used as the default phone device reference and its corresponding phone properties are inherited by the new Cisco Unified SIP IP phone.

Table 1 lists the names, types, and corresponding properties of supported phones that can be entered as values for the **device-reference** keyword. The description string configured with the device-name keyword is displayed as a help string when the new phone type is listed with the supported device types for the type (voice register pool) command.

The description string configured with the **device-name** keyword is displayed as a help string when the new phone type is listed with the supported device types for the type (voice register pool) command.

With respect to the **transport-type** keyword, most Cisco Unified SIP IP phones use UDP as the default transport protocol to connect to Cisco Unified CME while CiscoMobile-iOS and Jabber-Android use TCP. These configurations can be changed using the session-transport {udp | tcp} command in voice register pool or voice register template configuration mode.

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Example

The following example shows how to inherit the existing features of its phone model (9951) using the Fast-Track configuration approach. Phone model “9951” is used as the value of the reference-pooltype keyword. The maxNumCalls XML tag defines “3” as the maximum number of calls allowed per line while the busyTrigger XML tag defines "3" as the number of calls that triggers call forward busy per line on the phone.

```
voice register pool-type 9900
reference-pooltype 9951
device-name “SIP Phone 9900 addon module”
num-lines 24
addons 3
transport tcp
telnet
gsm-handoff
phoneload
xml-config maxNumCalls 3
xml-config busyTrigger 3
voice register pool 10
type 9900 addon 1 CKEM 2 CKEM 3 CKEM
id mac 1234.4567.7891
```

```
voice register global
mode cme
load 9900 P0S3-06-0-00
```

The following example shows how to inherit the existing features of its parent phone type (Cisco Unified 6921 SIP IP phone) using the Fast-Track configuration approach. Parent phone model “6921” is used as the value of the referencetype keyword.

```
voice register pool-type 6922
reference-pooltype 6921
device-name “SIP Phone 6922”

voice register pool 11
type 6922
id mac 1234.4567.7890
```

Related Commands

Command	Description
session-transport	Specifies the transport layer protocol that a SIP phone uses to connect to Cisco Unified CME.
type (voice register pool)	Defines a phone type for a SIP phone.

voice register session-server

To enter voice register session-server configuration mode to enable and configure a session manager in Cisco Unified CME for an external feature server, use the **voice register session-server** command in global configuration mode. To remove a session manager, use the **no** form of this command.

voice register session-server *session-server-tag*
no voice register session-server *session-server-tag*

Syntax Description

session-server-tag	Explicitly identifies a session manager for configuration tasks. Range is 1 to the maximum number of Cisco IP phones supported by a Cisco Unified CME router as set by the **max-ephones** command in telephony-service configuration mode.

Command Default

No session manager is created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW2	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.2(1)T	Cisco Unified CME 8.8	This command was modified to allow the maximum number of Cisco IP phones supported by a Cisco Unified CME router to be identified as the session manager.

Usage Guidelines

Provisioning and configuration information in the Cisco Unified Contact Center Express (Cisco Unified CCX) is automatically provided to Cisco United CME. Use the **voice register session-server** command to enter voice register session-server configuration mode and reconfigure and enable a session manager for Cisco Unified CCX on a Cisco Carrier Routing System when the configuration from Cisco Unified CCX is deleted or must be modified.

A single Cisco Unified CME can support multiple session managers.

After creating one or more session managers, use the **session-server** command in voice register pool configuration mode to identify a session manager for controlling a route point.

After creating one or more session managers, use the **session-server** command in ephone-dn configuration mode to specify session managers for monitoring a directory numbers.

Examples

The following is a partial output from the **show running-configuration** command, showing the configuration for the session manager, session-server 1:

```
!
```

```
voice register session-server 1
 keepalive 300
 register-id SB-SJ3-UCCX1_1164774025000
!
```

Related Commands

Command	Description
session-server	Specifies a session server to manage and monitor registration and subscription messages for an external feature server.

voice register template

To enter voice register template configuration mode and define a template of common parameters for SIP phones, use the **voice register template** command in global configuration mode. To remove a template, use the **no** form of this command.

voice register template *template-tag*
no voice register template *template-tag*

Syntax Description

template-tag	Declares a template tag. Range: 1 to 10.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	The maximum number of templates was increased from 5 to 10.
12.4(15)T	Cisco Unified CME 4.1	The increase in the template number was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Up to ten different templates can be defined and applied to SIP phones. You create the template with this command and then apply the template to a phone by using the **template** command in voice register pool configuration mode.

Examples

In the following example, template 1 is created by using the **voice register template** command.

```
Router(config)# voice register template 1
Router(config-register-temp)# anonymous block
Router(config-register-temp)# caller-id block
Router(config-register-temp)# voicemail 5001 timeout 15
```

Related Commands

	Description
anonymous block (voice register template)	Enables anonymous call blocking in a SIP phone template.
caller-id block (voice register template)	Enables caller-ID blocking for outbound calls from a specific SIP phone.
template (voice register pool)	Applies a template to a SIP phone.
voicemail (voice register template	Defines the extension that calls are forwarded to when an extension does not answer.

voice user-profile

To enter voice user-profile configuration mode and create a user profile for downloading by Extension Mobility for a particular individual phone user, use the **voice user-profile** command in global configuration mode. To delete an logout profile, use the **no** form of this command.

voice user-profile *profile-tag*
no voice user-profile *profile-tag*

Syntax Description

profile-tag	Unique number that identifies this profile during configuration tasks. Range: 1 to three times the maximum number supported phones, where maximum is platform and version dependent and defined by the **max-ephone** command.

Command Default

No user profile is created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to create a user profile containing a user's own personal settings, such as directory number, speed-dial lists, and services, for downloading to the IP phone when the individual phone user logs into a Cisco Unified IP phone that is registered in Cisco Unified CME and enabled for Extension Mobility.

Type **?** in voice profile configuration mode to see the commands that are available in this mode and that can be included in a user profile. The following example shows a list of commands that were available in voice user-profile configuration mode at the time that this document was written:

```
Router(config-user-profile)#?
Logout profile configuration commands:
 name    Define username and password for Extension Mobility.
 number    Create ip-phone line definition
 pin
 reset    Reset all phones associated with the profile being configured
 speed-dial    Define ip-phone speed-dial number
```

All directory numbers to be included in a default logout profile or voice-user profile must already be configured in Cisco Unified CME.

After creating or modifying a profile, use the **reset (voice user-profile)** command to reset all phones on which this profile is downloaded to propagate the modifications.

Examples

The following example shows the configuration for a voice-user profile to be downloaded when a phone user logs into a Cisco Unified IP phone that is enabled for Extension Mobility. The lines and speed-dial buttons in this profile that are configured on a phone after the user logs in depend on the phone type. For example, if the user logs into a Cisco Unified IP Phone 7970, all buttons are configured according to voice-user profile1. However, if the phone user logs into a Cisco Unified IP Phone 7960, all six lines are mapped to phone buttons and the speed dial is ignored because no button is available for speed dial.

```
pin 12345
user me password pass123
number 2001 type silent-ring
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for Extension Mobility and assigns a logout profile to this phone.
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones on which a particular logout profile or user profile is downloaded.

voice-class codec (voice register pool)

To assign a previously configured codec selection preference list, use the **voice-class codec** command in voice register pool configuration mode. To remove the codec preference assignment from the voice register pool, use the no form of this command.

voice-class codec *tag*
no voice-class codec

Syntax Description

tag	Unique number assigned to the voice class. Range is from 1 to 10000. The tag number maps to the tag number created by using the **voice class codec** command in dial-peer configuration mode.

Command Default

There is no codec preference assignment in the voice register pool configuration.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.

Usage Guidelines

During Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME) registration, a dial peer is created and that dial peer includes codec g729r8 by default. This command allows you to change the automatically selected default codec.

You can assign one voice class to each voice register pool. If you assign another voice class to a pool, the last voice class assigned replaces the previous voice class.

Note The **id** (voice register pool) command is required and must be configured before any other voice register pool commands. The **id** command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

Examples

The following partial sample output from the **show running-config** command shows that voice register pool 1 has been set up to use the previously configured codec voice class 1:

```
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
 proxy 10.2.161.187 preference 1 monitor probe icmp-ping
 alias 1 94... to 91011 preference 8
```

```
voice-class codec 1
```

Related Commands

	Description
codec (voice register pool)	Specifies the codec supported by a single Cisco SIP phone or a VoIP dial peer in a Cisco Unified SIP SRST or a Cisco Unified CME environment.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
voice class codec (dial-peer)	Assigns a previously configured codec selection preference list (codec voice class) to a VoIP dial peer.

voice-class mlpp (dial peer)

To assign a Multilevel Precedence and Preemption (MLPP) voice class to a POTS or VoIP dial peer, use the voice-class mlpp command in dial-peer configuration mode. To remove the voice class from the dial peer, use the **no** form of this command.

voice-class mlpp *tag*
no voice-class mlpp *tag*

Syntax Description

tag	Unique number that identifies the voice class. Range: 1 to 10000.

Command Default

The dial peer does not use an MLPP voice class.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

The voice class that you assign to the dial peer must first be configured using the voice class mlpp command in global configuration mode.

You can assign one voice class to each dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

Examples

The following example shows that VoIP dial peer 36 is assigned MLPP class 2.

```
Router(config)# dial-peer voice 36 voip
Router(config-dial-peer)# voice-class mlpp 2
```

Related Commands

Command	Description
service-domain (voice class)	Sets the service domain name in the MLPP voice class.
show dial-peer voice	Displays the configuration for all dial peers configured on the router.
voice class mlpp	Creates an MLPP voice class.

voice-class stun-usage

To configure voice class, enter voice class configuration mode called stun-usage and use the **voice-class stun-usage** command in global, dial-peer, ephone, ephone template, voice register pool, or voice register pool template configuration mode. To disable the voice class, use the **no** form of this command.

voice-class stun-usage *tag*
no voice-class stun-usage *tag*

Syntax Description

tag	Unique identifier in the range 1 to 10000.

Command Default

The voice class is not defined.

Command Modes

Global configuration (config)
Dial peer configuration (config-dial-peer)
Ephone configuration (config-ephone)
Ephone template configuration (config-ephone-template)
Voice register pool configuration (config-register-pool)
Voice register pool template configuration (config-register-pool)

Command History

Release	Cisco Product	Modification
12.4(22)T	Cisco Unified CME 7.0	This command was introduced.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. This command can be enabled in ephone summary, ephone template, voice register pool, or voice register pool template configuration mode.

Usage Guidelines

When the voice-class stun-usage is removed, the same is removed automatically from the dial-peer, ephone, ephone template, voice register pool, or voice register pool template configurations.

Examples

The following example shows how to set the **voice class stun-usage** tag to 10000:

```
Router(config)# voice class stun-usage 10000
Router(config-ephone)# voice class stun-usage 10000
Router(config-voice-register-pool)# voice class stun-usage 10000
```

Related Commands

Command	Description
stun usage firewall-traversal flowdata	Enables firewall traversal using STUN.
stun flowdata agent-id	Configures the agent ID.

voice-gateway system

To enter voice-gateway configuration mode and create a voice gateway configuration, use the **voice-gateway system** command in global configuration mode. To remove the configuration, use the **no** form of this command.

voice-gateway system *tag*
no voice-gateway system *tag*

Syntax Description

tag	Unique number that identifies the voice gateway. Range: 1 to 25. There is no default value.

Command Default

Gateway configuration is not defined.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command has been integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command enters voice-gateway configuration mode to define the parameters for a voice gateway using the auto-configuration feature. Define a configuration for each Cisco voice gateway whose analog FXS ports you want under the control of this Cisco Unified CME router.

Examples

The following example shows a voice gateway configuration:

```
voice-gateway system  1
 network-locale FR
 type VG224
 mac-address 001F.A30F.8331
 voice-port 0-23
 create cnf-files
```

Related Commands

Command	Description
mac-address	Defines the MAC address of the Cisco voice gateway that downloads its configuration from Cisco Unified CME.
type	Defines the type of voice gateway to autoconfigure in Cisco Unified CME.
voice-port	Identifies the analog ports on the voice gateway that register to Cisco Unified CME.

voicemail (telephony-service)

To define the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed, use the **voicemail** command in telephony-service configuration mode. To disable the Messages button, use the **no** form of this command.

voicemail *phone-number*
no voicemail

Syntax Description

phone-number	Phone number that is configured as a speed-dial number for retrieving messages.

Command Default

No phone number is configure and the Messages button is disabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

This command configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed. The same telephone number is configured for voice messaging for all Cisco IP phones connected to the router.

Examples

The following example sets the phone number 914085550100 as the speed-dial number that is dialed to retrieve messages when the Messages button is pressed:

```
Router(config)# telephony-service
Router(config-telephony)# voicemail 914085550100
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.
vm-device-id (ephone)	Defines the voice-mail ID string.

voicemail (voice register global)

To define the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed, use the **voicemail** command in voice register global configuration mode. To disable the Messages button, use the **no** form of this command.

voicemail *phone-number*
no voicemail

Syntax Description

phone-number	Telephone number that is speed-dialed for retrieving messages.

Command Default

No phone number is configure and the Messages button is disabled.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed. The same telephone number is configured for voice messaging for all Cisco IP phones connected to the router.

Examples

The following example shows how to set telephone number 914085550100 as the speed-dial number to retrieve messages when the Messages button is pressed:

```
Router(config)# voice register global
Router(config-register-global)# voicemail 914085550100
```

Related Commands

	Description
url (voice register global)	Provision uniform resource locators (URLs) for feature buttons on Cisco IP phones.
voicemail (voice register template)	Defines the extension that calls are forwarded to when an extension does not answer.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

voicemail (voice register template)

To define the extension that calls are forwarded to when an extension does not answer, use the **voicemail** command in voice register template configuration mode. To disable the voicemail extension, use the **no** form of this command.

voicemail *phone-number* **timeout** *timeout*
no voicemail

Syntax Description

phone-number	Telephone number to which calls are forwarded when an extension does not answer.
timeout *seconds*	Duration that a call can ring with no answer before the call is forwarded to the voicemail extension. Range is 5 to 60000. There is no default value.

Command Default

This command has no default behavior or values.

Command Modes

Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.

Usage Guidelines

This command defines the destination extension for voicemail when an extension on a SIP phone does not answer. To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following example shows how to set telephone number 914085550100 as the number to be dialed to retrieve messages when the Messages button is pressed:

```
Router(config)# voice register template 1
Router(config-register-temp)# voicemail 50100 timeout 15
```

Related Commands

	Description
template (voice register pool)	Applies a template to a SIP phone.
url (voice register global)	Provisions uniform resource locators (URLs) for feature buttons on Cisco IP phones.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

	Description
voicemail (voice register global)	Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.

voice-port (voice-gateway)

To identify the analog ports on the voice gateway that register to Cisco Unified CME, use the **voice-port** command in voice-gateway configuration mode. To remove the ports, use the **no** form of this command.

voice-port *port-range*
no voice-port

Syntax Description

port-range	Individual port number, or range of port numbers, on the voice gateway controlled by Cisco Unified CME. Enter individual port values separated by a comma (,) or enter a range using a hyphen (x-y). There is no default value.

Command Default

No voice ports are supported.

Command Modes

Voice-gateway configuration (config-voice-gateway)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified CME 7.1	This command was introduced.
12.4(24)T	Cisco Unified CME 7.1	This command has been integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command sets the total number of analog endpoints on the voice gateway that you intend to register to the Cisco Unified CME router. The Cisco VG202 supports two ports, Cisco VG204 supports four ports, and the Cisco VG224 supports 24 ports, numbered 0 to 23.

Examples

The following example shows a configuration for a Cisco VG224 voice gateway with 24 ports:

```
voice-gateway system  1
 network-locale FR
 type VG224
 mac-address 001F.A30F.8331
 voice-port 0-23
 create cnf-files
```

Related Commands

Command	Description
network-locale (voice-gateway)	Selects a geographically specific set of tones and cadences for the voice gateway's analog endpoints that register to Cisco Unified CME.
type (voice-gateway)	Defines the type of voice gateway to autoconfigure in Cisco Unified CME.

vpn-gateway

To enter vpn-gateway url, use the vpn-gateway command in vpn-group configuration mode. To disable the vpn-gateway configuration, use the **no** form of this command.

vpn-gateway **number** [**url**]
no **vpn-group**

Syntax Description

number	Vpn-gateway numbers. Range: 1-3.
url	VPN concentrator address url as https://<IP>/policy.

Command Default

vpn-gateway is not configured.

Command Modes

Vpn-group configuration (conf-vpn-group).

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to enter vpn-gateway urls. You can define up to 3 vpn-gateways urls for SSLVPN phones.

Examples

The following example shows vpn-gateway 1 configured for vpn-group 1:

```
Router# show run
!
!
!
voice-card 3
 dspfarm
 dsp services dspfarm
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  host-id-check disable
 sip
```

Related Commands

Command	Description
vpn-group	Specifies a vpn-group.
vpn-trustpoint	Specifies a vpn-gateway trustpoint.

vpn-group

To enter vpn-group mode, use the vpn-group command in voice service voip configuration mode. To delete all configurations associated with a vpn-group, use the **no** form of this command.

vpn-group **tag**
no **vpn-group**

Syntax Description

tag	Vpn-group tag number. Range: 1-2.

Command Default

vpn-group is not configured.

Command Modes

Voice service voip (conf-voi-serv)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to create vpn-groups. A vpn-group is a redundancy ordered list of up to 3 vpn-gateways that an SSL VPN client on a phone can connect to. You can create 2 vpn-groups.

Examples

The following example shows vpn-group 1:

```
Router# show run
!
!
!
voice-card 3
 dspfarm
 dsp services dspfarm
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  host-id-check disable
 sip
```

Related Commands

Command	Description
vpn-gateway	Specifies a vpn-gateway URL.
vpn-trustpoint	Specifies a vpn-gateway trustpoint.
vpn-hash-algorithm	Specifies vpn hash encryption for the trustpoints.

vpn-hash-algorithm

To specify the algorithm to hash the VPN certificate provided in the configuration file downloaded to the phone, use the vpn-hash-algorithm command in vpn-group configuration mode. To disable vpn-hash-encryption, use the **no** form of this command.

vpn-hash-algorithm sha-1
no vpn-hash-algorithm

Syntax Description

sha-1	Encryption algorithm.

Command Default

vpn-hash-algorithm is not configured

Command Modes

Vpn-group configuration (conf-vpn-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to specify the algorithm to hash the VPN certificate provided in the configuration file downloaded to the phone.

Examples

The following example shows vpn-hash-algorithm configured in vpn-group 1:

```
Router# show run
!
!
!
voice-card 3
 dspfarm
 dsp services dspfarm
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  host-id-check disable
 sip
```

Related Commands

Command	Description
vpn-group	Specifies a vpn-group.
vpn-trustpoint	Specifies a vpn-gateway trustpoint.

vpn-profile

To enter vpn-profile mode to configure vpn-profiles in Cisco Unified CME, use the **vpn-profile** command in voice service voip configuration mode. To remove the entire vpn-profile configuration, use the no form of this command.

vpn-profile *tag*
no vpn-profile

Syntax Description

tag	Vpn-profile tag number. Range:1-6,

Command Default

No vpn-profile is configured.

Command Modes

Voice service voip (conf-voi-serv)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to create one or more vpn-profiles on Cisco Unified CME. You can create 6 vpn-profiles.

Examples

The following example shows 3 vpn-profiles configured:

```
Router# show run
!
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  keepalive 50
  auto-network-detect enable
  host-id-check disable
 vpn-profile 2
  mtu 1300
  password-persistent enable
  host-id-check enable
 vpn-profile 4
  fail-connect-time 50
 sip
!
!
```

Related Commands

Command	Description
voice-service-voip	Enters voice-service configuration mode for Voice Over IP (VoIP) encapsulation.

Command	Description
vpn-group	Enters vpn-group configuration mode.

vpn-trustpoint

To configure a vpn gateway trustpoint, use the vpn-trustpoint command in vpn-group configuration mode. To disable a vpn-gateway trustpoint associated with a vpn-group, use the **no** form of this command.

vpn-trustpoint number [{**raw** | **trustpoint**}] **word** [{**leaf** | **root**}]
no vpn-trustpoint

Syntax Description

number	Number of allowed trustpoints. Range is from 1 to 10.
raw	(Optional) Allows to enter VPN Gateway Trustpoint in raw form.
trustpoint	(Optional) Allows to enter VPN Gateway Trustpoint in IOS format.
leaf	Get the 1st leaf cert of the Trustpoint.
root	Get the root cert of the Trustpoint.

Command Default

vpn-trustpoint is not configured.

Command Modes

Vpn-group (conf-vpn-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5	This command was introduced.

Usage Guidelines

Use this command to create vpn-trustpoints for a vpn-group. You can configure as many as 10 vpn-trustpoints in a vpn-group. All vpn trustpoints must be entered in either raw or trustpoint (IOS) format.

Examples

The following example shows vpn-trustpoint 1 entered in trustpoint (IOS) format:

```
Router# show run
!
!
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 vpn-group 1
  vpn-gateway 1 https://9.10.60.254/SSLVPNphone
  vpn-trustpoint 1 trustpoint cme_cert root
  vpn-hash-algorithm sha-1
 vpn-profile 1
  host-id-check disable
 sip
```

Related Commands

Command	Description
vpn-grouptrustpoint	Defines a vpn-group.

Cisco Unified CME Commands: W

web admin customer

To define a username and password for a Cisco Unified CME customer administrator, use the **web admin customer** command in telephony-service configuration mode. To disable a customer administrator login, use the **no** form of this command.

web admin customer name *username* {**password** *string* | **secret** {**0** | **5**} *string*}
no web admin customer

Syntax Description

name *username*	Username for the customer administrator. String can contain a maximum of 28 alphanumeric characters. Default is Customer.
password *string*	Character string for login authentication, which will be stored in the running configuration as plain text. String can contain a maximum of 28 alphanumeric characters. Default is no password.
secret {**0** | **5**} *string*	Character string for login authentication, which will be stored in the running configuration as encrypted using Message Digest 5 (MD5). The digit 0 or 5 specifies whether the displayed string that follows is encrypted: • **0**—Password that follows is not encrypted. • **5**—Password that follows is encrypted.

Command Default

Default is a customer administrator with username Customer and no password.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Usage Guidelines

This command enables customer administrator access for the Cisco Unified CME graphical user interface (GUI).

Examples

The following example defines a customer administrator named user22 whose password is pw567890:

```
Router(config)# telephony-service
Router(config-telephony)# web admin customer name user22 password pw567890
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.
web customize load	Loads and parses an XML file in router flash memory to customize a GUI for a customer administrator.

web admin system

To define a username and password so that a system administrator can log in to the Cisco Unified CME router through a web browser, use the **web admin system** command in telephony-service configuration mode. To disable a system administrator login, use the **no** form of this command.

web admin system [**name** *username*] [{**password** *string* | **secret** {**0** | **5**} *string*}]
no web admin system

Syntax Description

name username	(Optional) Unique alphanumeric string to identify a user for this authentication credential only. String can contain a maximum of 28 alphanumeric characters. Default name is Admin.
password *string*	(Optional) Unique alphanumeric string to be used by the system administrator which will be stored in the running configuration as plain text. This password is typically known by more than one administrator user and may be created for the default system administrator username. String can contain a maximum of 28 alphanumeric characters. Default is no password.
secret {**0** | **5**} *string*	(Optional) Unique alphanumeric string for a secret password which will be stored in the running configuration as unencrypted plain text or as encrypted using Message Digest 5 (MD5). This password is typically known by only a select number of administrator users. • **0**—Save *string* as unencrypted plain text in running configuration.. • **5**—Save encrypted *string* in running configuration.

Command Default

Default is a system administrator with username Admin and no password.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Usage Guidelines

This command enables system administrator access for the Cisco Unified CME graphical user interface (GUI).

The user name parameter of any authentication credential must be unique. Do not use the same value for a user name when you configure any two or more authentication credentials in Cisco Unified CME, such as the username for any Cisco United CME GUI account and the user name in a profile for Extension Mobility.

Use the **secret 5** keyword pair to instruct the system to encrypt the system administrator password with MD5 and to save the encrypted version in the running configuration.

Examples

The following example establishes a system administrator named user1 whose secret password will be encrypted in the running configuration:

```
Router(config)# telephony-service
Router(config-telephony)# web admin system name user1 secret 5 pw234567
```

An encrypted version of the preceding string is saved in the running configuration, as shown in the following partial example. The digit 5 that appears after the **secret** keyword in the running configuration indicates that the password that follows is shown in its encrypted version.

```
Router(config)# show running-config
!
!
!
web admin system name user1 secret 5 $1$TCyK$OU/NSQ/VtAU2ibHdi8Uau
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.

web customize load

To load and parse an eXtensible Markup Language (XML) file in router flash memory to customize a Cisco CallManager Express graphic user interface (GUI) for a customer administrator, use the **web customize load** command in telephony-service configuration mode. To disable the customized GUI and use the system administrator GUI for the customer administrator, use the **no** form of this command.

web customize load *filename*
no web customize load

Syntax Description

filename	Name of the XML file in router flash memory that defines the customer administrator GUI.

Command Default

The standard system administrator GUI is used.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Usage Guidelines

Use this command with Cisco ITS V2.1and later versions.

Examples

The following example specifies a file named cust_admin_gui.xml as the file that defines the GUI for Cisco CME customer administrators:

```
Router(config)# telephony-service
Router(config-telephony)# web customize load cust_admin_gui.xml
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.

Cisco Unified CME Commands: X

xml-config

To define the phone-specific XML tags that can be used in the configuration file, use the **xml-config** command in the voice register pooltype mode. To remove the XML tags, use the **no** form of this command.

xml-config [{**maxNumcalls** **maxNumCalls** | **busyTrigger** **busyTrigger** | **custom** **custom**}]
no **xml-config** [{**maxNumcalls** **maxNumCalls** | **busyTrigger** **busyTrigger** | **custom** **custom**}]

Syntax Description

maxNumcalls	Defines the maximum number of calls allowed per line.
busyTrigger	Defines the number of calls that triggers call forward busy per line on the SIP phone.
custom	Defines the custom XML tags that can be appended at the end of the phone-specific CNF configuration profile using the custom option.

Command Default

The phone-specific XML tags are not defined.

Note When the reference-pooltype command is configured, the XML configuration value of the reference phone is inherited.

Command Modes

Voice Register Pool Configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to define the phone specific XML tags that can be used in the configuration file. the maximum nunber of call allowed per line and the number of call that triggers call forward busy per line information will be used while generating the XML file.

Examples

The following example shows how to define the phone specific XML tags that can be used in the configuration file:

```
Router# configure terminal
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# xml-config maxNumCalls 3
Router(config-register-pool-type)# xml-config busyTrigger 3
Router(config-register-pool-type)# xml-config custom <custom-sftp>1</custom-sftp>
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.
phoneload-support	Enables support for phone loads.

xmlschema

Effective with Cisco Unified CME 4.0, the **xmlschema** command was made obsolete.

For earlier releases, to specify the URL for a Cisco CME eXtensible Markup Language (XML) application program interface (API) schema, use the **xmlschema** command in telephony-service configuration mode. To set the URL for the XML API schema to the default, use the **no** form of this command.

xmlschema *schema-url*
no xmlschema

Syntax Description

schema-url	Local or remote URL as defined in RFC 2396.

Command Default

Url for Cisco XML API schema is srst-its.xsd.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made obsolete.
12.4(9)T	Cisco Unified CME 4.0	This command was made obsolete in Cisco IOS Release 12.4(9)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Examples

The following example specifies a URL for an XML API schema:

```
Router(config)# telephony-service
Router(config-telephony)# xmlschema http://server2.example.com/schema/schema1.xsd
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.

xmltest

Effective with Cisco Unified CME 4.0, the **xmltest** command was made obsolete.

For earlier releases, to specify that the HTTP payload in eXtensible Markup Language (XML) application program interface (API) queries be interpreted as having form format, use the **xmltest** command in telephony-service configuration mode. To specify that the HTTP payload should be interpreted as plain text (no form) format, use the **no** form of this command.

xmltest
no xmltest

Syntax Description

This command has no arguments or keywords.

Command Default

Default format is plain text (no form) format.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)Ts	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made obsolete.
12.4(9)T	Cisco Unified CME 4.0	This command was made obsolete in Cisco IOS Release 12.4(9)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Examples

The following example specifies that the HTTP payload in XML API queries be interpreted as having form format:

```
Router(config)# telephony-service
Router(config-telephony)# xmltest
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.

xmlthread

Effective with Cisco Unified CME 4.0, the **xmlthread** command was made obsolete.

For earlier releases, to set the maximum number of concurrent Cisco CME eXtensible Markup Language (XML) application program interface (API) queries, use the **xmlthread** command in telephony-service configuration mode. To set the maximum number of queries to the default, use the **no** form of this command.

xmlthread *number*
no xmlthread

Syntax Description

number	Maximum number of XML API queries. Range is from 1 to 5. Default is 2.

Command Default

The maximum number of queries is 2.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made obsolete.
12.4(9)T	Cisco Unified CME 4.0	This command was made obsolete in Cisco IOS Release 12.4(9)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Examples

The following example sets the maximum number of XML API queries to 5:

```
Router(config)# telephony-service
Router(config-telephony)# xmlthread 5
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.

xml user

To define a user who is authorized to use XML applications to execute commands, use the **xml user** command in telephony-service configuration mode. To delete the user, use the **no** form of this command.

xml user *user-name* **password [0|6]** *password privilege-level*
no xml user *user-name* **password** *password privilege-level*

Syntax Description

user-name	Unique string used by authorized user to access Cisco Unified CME. Maximum length of string: 19 alphanumeric characters.
password *password*	Alphanumeric string to be used with this user name to provide access to Cisco Unified CME. Maximum length of string: 19 alphanumeric characters.
privilege-level	Level of access to Cisco IOS commands to be granted to this user. Only the commands with the same or a lower level can be executed via XML. Range is 0 to 15.

Command Default

User name is not defined.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command was enhanced for password encryption, based on Unified CME password policy.

Usage Guidelines

This command creates a credential be used by an authorized user to access Cisco Unified CME via XML and enable the user to execute all the Cisco IOS commands associated with a particular privilege level.

To change the default privilege level for one or more Cisco IOS commands, use the **privilege** command in global configuration mode.

From Unified CME 12.6 onwards, you must configure password encryption using the parameters **[0|6]**. This in accordance with Unified CME Password Policy. The 0 in the parameter [0|6] mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

Examples

The following example defines user23 as an authorized user at level 15:

```
Router(config)# telephony-service
Router(config-telephony)# xml user user23 password 3Rs92uzQ 15
```

Related Commands

Command	Description
privilege	Configures a new privilege level for users and associates commands with that privilege level.