O

offer call-hold

To specify globally how the POTS-SIP gateway should initiate call-hold requests, use the **offer call-hold** command in SIP user-agent configuration mode or voice class tenant configuration mode. To disable a method of initiating call hold, use the **no** form of this command.

offer call-hold {**conn-addr** | **direction-attr** | **system**}
no offer call-hold {**conn-addr** | **direction-attr** | **system**}

Syntax Description

conn-addr	Specifies the RFC 2543 method of using the connection address for initiating call-hold requests. The RFC 2543 method uses 0.0.0.0.
direction-attr	Specifies the current RFC 3264 method of using the direction attribute (a=sendonly) for initiating call-hold requests.
system	Specifies how the call-hold requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations

Command Default

direction-attr

Command Modes

SIP user-agent configuration

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.3(8)T	This command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**.

Usage Guidelines

Cisco POTS-SIP gateways support receiving call-hold requests in either of the two formats, but the direction attribute is recommended. Specifying a call-hold format is only available globally with the **offer call-hold** command; configuration is not available at the dial-peer level.

Examples

The following example initiates call hold by configuring the gateway to send a=sendonly in the Session Description Protocol (SDP). Using the **direction-attr**keyword is the current and preferred method to initiate call hold.

```
sip-ua
 retry invite 3
 offer call-hold direction-attr
```

The following example initiates call hold by configuring the gateway to send 0.0.0.0 as the IP address in the c=line.

```
sip-ua
 retry invite 3
 offer call-hold conn-addr
```

The following example initiates call hold by configuring the gateway in the voice class tenant configuration mode:

```
Router(config-class)# offer call-hold system
```

Related Commands

Command	Description
show sip-ua status	Displays status for the SIP UA.
suspend-resume	Enables SIP Suspend and Resume functionality.

operation

To select a specific cabling scheme for E&M ports, use the **operation**command in voice-port configuration mode. To restore the default, use the **no** form of this command.

operation {**2-wire** | **4-wire**}
no operation {**2-wire** | **4-wire**}

Syntax Description

2 -wire	Two-wire E&M cabling scheme.
4 -wire	Four-wire E&M cabling scheme.

Command Default

2-wire E&M cabling scheme

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(1)MA	This command was implemented on the Cisco MC3810.

Usage Guidelines

This command affects only voice traffic. Signaling is independent of 2-wire versus 4-wire settings. If the wrong cable scheme is specified, the user might get voice traffic in only one direction.

Using this command on a voice port changes the operation of both voice ports on a VPM card. The voice port must be shut down and then opened again for the new value to take effect.

This command is not applicable to FXS or FXO interfaces because they are, by definition, 2-wire interfaces.

Examples

The following example specifies that an E&M port uses a 4-wire cabling scheme:

```
voice-port 1/0/0
 operation 4-wire
```

The following example specifies that an E&M port uses a 2-wire cabling scheme:

```
voice-port 1/1
 operation 2-wire
```

options-ping

To enable in-dialog OPTIONS, use the **options-ping** command in global configuration mode or voice class tenant configuration mode. To disable, use the **no** form of this command.

options-ping *seconds* **[system]**
no options-ping *seconds* **[system]**

Syntax Description

seconds	Intervals, in seconds OPTIONS transactions are sent. Range is 60-1200, there is no default.
system	Specifies that the in-dialog OPTIONS, use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations

Command Default

This command is disabled by default.

Command Modes

Global

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(11)T	This command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

The in-dialog OPTIONS refresh command eanbles an alterante refresh mechanism to RTP/RTCP media inactivity timer and session timer can be used on SIP-to-SIP and SIP-to-H.323 calls. The refresh with in-dialog OPTIONS method is meant to only be hop-to-hop, and not end-to-end. Since session timer achieves similar results, the OPTIONs refresh/ping will not take affect when session timer is negotiated. The behavior on the H.323 endpoint is as if it was a TDM-SIP call. The generating in-dialog OPTIONS is enabled at the global level or dialpeer level. The system default setting is disabled. This feature can be use by both a TDM voice gateway and an IP-to-IP gateway.

Examples

The following example sets the in-dialog refresh time to 60 seconds:

```
Router(conf-serv-sip)# options-ping
```

The following example sets the in-dialog refresh time in the voice class tenant configuration mode:

```
Router(conf-class)# options-ping system
```

Related Commands

Command	Description
options-ping	Enables in-dialog OPTIONS at the global level.
options-ping (dial peer)	Enables in-dialog OPTIONS on a dial-peer.

options-ping (dial-peer)

To enable in-dialog OPTIONS, use the **options-ping** command in global configuration mode. To disable, use the **no** form of this command.

options-ping *seconds*
no options-ping *seconds*

Syntax Description

seconds	Intervals, in seconds OPTIONS transactions are sent. Range is 60-1200, there is no default.

Command Default

This command is disabled by default.

Command Modes

dial peer configuration mode

Command History

Release	Modification
12.4(11)T	This command was introduced.

Usage Guidelines

The in-dialog OPTIONS refresh command eanbles an alterante refresh mechanism to RTP/RTCP media inactivity timer and session timer can be used on SIP-to-SIP and SIP-to-H.323 calls. The refresh with in-dialog OPTIONS method is meant to only be hop-to-hop, and not end-to-end. Since session timer achieves similar results, the OPTIONs refresh/ping will not take affect when session timer is negotiated. The behavior on the H.323 endpoint is as if it was a TDM-SIP call. The generating in-dialog OPTIONS is enabled at the global level or dialpeer level. The system default setting is disabled. This feature can be use by both a TDM voice gateway and an IP-to-IP gateway.

Examples

The following example sets the in-dialog refresh time to 60 seconds:

```
Router(conf-serv-sip)# options-ping 60
```

Related Commands

Command	Description
options-ping	Enables in-dialog OPTIONS at the global level.
options-ping (dial peer)	Enables in-dialog OPTIONS on a dial-peer.

outbound-proxy

To configure a Session Initiation Protocol (SIP) outbound proxy for outgoing SIP messages globally on a Cisco IOS voice gateway, use the **outbound-proxy** command in voice service SIP configuration mode or voice class tenant configuration mode. To globally disable forwarding of SIP messages to a SIP outbound proxy globally, use the **no** form of this command.

outbound-proxy {**dhcp** | **ipv4:***ip-address*[{**:***port-number* | **dns:***host***:***domain* [{**reuse**}]}]} **[system]**
no outbound-proxy

Syntax Description

dhcp	Specifies the SIP outbound proxy globally for a Cisco IOS voice gateway; all SIP dialog-initiating requests are sent to the SIP server obtained via DHCP.
ipv4 **:** *ip-address*	Specifies the SIP outbound proxy globally for a Cisco IOS voice gateway; all SIP dialog-initiating requests are sent to this IP address. The colon is required.
: *port-number*	(Optional) The port to which all SIP dialog-initiating requests are sent at the specified IP address. Port number ranges from 0 to 65535. The default is 5060. The colon is required.
dns **:** *host* **:** *domain*	Specifies the SIP outbound proxy globally for a Cisco IOS voice gateway; all initiating requests are sent to the specified destination domain. The colon is required.
reuse	(Optional) Reuses the outbound proxy address established during registration for all subsequent registration refreshes and calls.
system	Specifies that the outbound proxy for outgoing SIP messages use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations

Command Default

The Cisco IOS voice gateway does not forward outbound SIP messages to a proxy.

Command Modes

Voice service VoIP SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

Command History

Release	Modification
12.4(15)T	This command was introduced.
12.4(22)T	Support for IPv6 was added.
12.4(22)YB	This command was modifed. The **dhcp** keyword was added.
15.0(1)M	This command was integrated in Cisco IOS Release 15.0(1)M.
15.1(2)T	This command was modified. The **reuse** keyword was added.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**.

Release	Modification
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

You can use the **outbound-proxy** command in voice service SIP configuration mode to specify outbound proxy settings globally for a Cisco IOS voice gateway. You can also use the **voice-class sip outbound-proxy** command in dial peer voice configuration mode to configure settings for an individual dial peer that override or defer to the global settings for the gateway. However, if both a Cisco Unified Communications Manager Express (CME) and a SIP gateway are configured on the same router, then there is a scenario that can cause incoming SIP messages from line-side phones to be confused with SIP messages coming from the network side. To avoid failed calls caused by this scenario, disable the SIP outbound proxy setting for all line-side phones on a dial peer using the **outbound-proxy system** command in voice register global configuration mode.

Examples

The following example shows how to specify the SIP outbound proxy globally for a Cisco IOS voice gateway using an IP address:

```
Router> enable
Router# configure
 terminal
Router(config)# voice
 service
 voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# outbound
-proxy
 ipv4
:10.1.1.1
```

The following example shows how to specify the SIP outbound proxy globally for a Cisco IOS voice gateway using a destination hostname and domain:

```
Router> enable
Router# configure
 terminal
Router(config)# voice
 service
 voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# outbound
-proxy
 dns:sipproxy:example.com
```

The following example shows how to specify the SIP outbound proxy globally for a Cisco IOS voice gateway using the DHCP protocol:

```
Router> enable
Router# configure
 terminal
Router(config)# voice
 service
 voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# outbound
-proxy
 dhcp
```

The following example shows how to specify the SIP outbound proxy globally in the voice class tenant configuration mode:

```
Router(config-class)# outbound-proxy system
```

Related Commands

Command	Description
outbound-proxy system	Specifies whether Cisco Unified CME line-side SIP phones use the outbound proxy settings configured globally for a Cisco IOS voice gateway.
voice-class sip outbound-proxy	Configures SIP outbound proxy settings for an individual dial peer that override global settings for the Cisco IOS voice gateway.

outbound retry-interval

To define the retry period for attempting to establish the outbound relationship between border elements, use the **outbound retry-interval** command in Annex G neighbor service configuration mode. To disable the command, use the **no** form of this command.

outbound retry-interval *interval*
no outbound retry-interval

Syntax Description

interval	Amount of time, in seconds, to establish the outbound relationship. Range is from 1 to 2147483. The default is 30.

Command Default

30 seconds

Command Modes

Annex G neighbor service configuration (config-nxg-neigh-svc)

Command History

Release	Modification
12.2(11)T	This command was introduced.

Usage Guidelines

Service relationships are defined to be unidirectional. When a service relationship is established between border element A and border element B, A is entitled to send requests to B and expect responses. For B to send requests to A and expect responses, a second service relationship must be established. From A’s perspective, the service relationship it establishes with B is designated as the "outbound" service relationship.

Use this command to set the retry period for attempting to bring up the outbound relationship between border elements.

Examples

The following example shows how to set the retry interval to 300 seconds (5 minutes):

```
Router(config-nxg-neigh-svc)
#
outbound retry-interval 300
```

Related Commands

Command	Description
access -policy	Requires that a neighbor be explicitly configured.
inbound ttl	Sets the inbound time-to-live value.
retry interval	Defines the time between delivery attempts.
retry window	Defines the total time that a border element will attempt delivery.
service -relationship	Establishes a service relationship between two border elements.
shutdown	Enables or disables the border element.

outgoing called-number

To configure debug filtering for outgoing called numbers, use the outgoing called-number command in call filter match list configuration mode. To disable, use the **no** form of this command.

outgoing called-number *string*
no outgoing called-number *string*

Syntax Description

string	Series of digits that specify a pattern for the E.164 or private dialing plan telephone number. Valid entries are the digits 0 to 9, the letters A to D, and the following special characters: • The asterisk (*) and pound sign (#) that appear on standard touchtone dial pads. On the Cisco 3600 series routers only, these characters cannot be used as leading characters in a string (for example, *650). • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit (this character is used as a wildcard). On the Cisco 3600 series routers, the period cannot be used as a leading character in a string (for example, .650). • Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage. • Plus sign (+), which indicates that the preceding digit occurred one or more times. **Note** The plus sign used as part of a digit string is different from the plus sign that can be used in front of a digit string to indicate that the string is an E.164 standard number. • Circumflex (^), which indicates a match to the beginning of the string. • Dollar sign ($), which matches the null string at the end of the input string. • Backslash symbol (\), which is followed by a single character; matches that character. Can be used with a single character with no other significance (matching that character). • Question mark (?), which indicates that the preceding digit occurred zero or one time. • Brackets ([ ]), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters 0 to 9 are allowed in the range. • Parentheses ( ), which indicate a pattern and are the same as the regular expression rule.

Command Default

No default behavior or values

Command Modes

Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Usage Guidelines

The outgoing called number goes out after number translation and expansion.

Examples

The following example shows the voice call debug filter set to match outgoing called number 8288807:

```
call filter match-list 1 voice
 outgoing called-number 8288807
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
incoming called-number (call filter match list)	Configure debug filtering for incoming called numbers.
incoming calling-number	Configure debug filtering for incoming calling numbers.
incoming dialpeer	Configure debug filtering for the incoming dial peer.
incoming secondary-called-number	Configure debug filtering for incoming called numbers from the second stage of a two-stage scenario.
outgoing calling-number	Configure debug filtering for outgoing calling numbers.
outgoing dialpeer	Configure debug filtering for the outgoing dial peer.
show call filter match-list	Display call filter match lists.

outgoing calling-number

To configure debug filtering for outgoing calling numbers, use the outgoing calling-number command in call filter match list configuration mode. To disable, use the **no** form of this command.

outgoing calling-number *string*
no outgoing calling-number *string*

Syntax Description

string	Series of digits that specify a pattern for the E.164 or private dialing plan telephone number. Valid entries are the digits 0 to 9, the letters A to D, and the following special characters: • The asterisk (*) and pound sign (#) that appear on standard touchtone dial pads. On the Cisco 3600 series routers only, these characters cannot be used as leading characters in a string (for example, *650). • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit (this character is used as a wildcard). On the Cisco 3600 series routers, the period cannot be used as a leading character in a string (for example, .650). • Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage. • Plus sign (+), which indicates that the preceding digit occurred one or more times. **Note** The plus sign used as part of a digit string is different from the plus sign that can be used in front of a digit string to indicate that the string is an E.164 standard number. • Circumflex (^), which indicates a match to the beginning of the string. • Dollar sign ($), which matches the null string at the end of the input string. • Backslash symbol (\), which is followed by a single character; matches that character. Can be used with a single character with no other significance (matching that character). • Question mark (?), which indicates that the preceding digit occurred zero or one time. • Brackets ([ ]), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters 0 to 9 are allowed in the range. • Parentheses ( ), which indicate a pattern and are the same as the regular expression rule.

Command Default

No default behavior or values

Command Modes

Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Usage Guidelines

The outgoing calling number goes out after number translation and expansion.

Examples

The following example shows the voice call debug filter set to match outgoing calling number 5550124:

```
call filter match-list 1 voice
 outgoing calling-number 5550124
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
incoming called-number (call filter match list)	Configure debug filtering for incoming called numbers.
incoming calling-number	Configure debug filtering for incoming calling numbers.
incoming dialpeer	Configure debug filtering for the incoming dial peer.
incoming secondary-called-number	Configure debug filtering for incoming called numbers from the second stage of a two-stage scenario.
outgoing called-number	Configure debug filtering for outgoing called numbers.
outgoing dialpeer	Configure debug filtering for the outgoing dial peer.
show call filter match-list	Display call filter match lists.

outgoing dialpeer

To configure debug filtering for the outgoing dial peer, use the **outgoing dialpeer** command in call filter match list configuration mode. To disable, use the **no** form of this command.

outgoing dialpeer *tag*
no outgoing dialpeer *tag*

Syntax Description

tag	Digits that identify a specific dial peer. Valid entries are 1 to 2,147,483,647.

Command Default

No default behavior or values

Command Modes

Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example shows the voice call debug filter set to match outgoing dial peer 12:

```
call filter match-list 1 voice
 outgoing dialpeer 12
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
incoming called-number (call filter match list)	Configure debug filtering for incoming called numbers.
incoming calling-number	Configure debug filtering for incoming calling numbers.
incoming dialpeer	Configure debug filtering for the incoming dial peer.
incoming port	Configure debug filtering for the incoming port.
outgoing called-number	Configure debug filtering for outgoing called numbers.
outgoing calling-number	Configure debug filtering for outgoing calling numbers.
outgoing port	Configure debug filtering for the outgoing port.
show call filter match-list	Display call filter match lists.

outgoing media local ipv4

To configure debug filtering for the outgoing media local IPv4 addresses for the voice gateway receiving the media stream, use the outgoing media local ipv4 command in call filter match list configuration mode. To disable, use the **no** form of this command.

outgoing media local ipv4 *ip_address*
no outgoing media local ipv4 *ip_address*

Syntax Description

ip_address	IP address of the local voice gateway

Command Default

No default behavior or values

Command Modes

Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example shows the voice call debug filter set to match outgoing media on the local voice gateway, which has IP address 192.168.10.255:

```
call filter match-list 1 voice
 outgoing media local ipv4 192.168.10.255
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
incoming media local ipv4	Configure debug filtering for the incoming media IPv4 addresses for calls to the IP side from the local voice gateway.
incoming media remote ipv4	Configure debug filtering for the incoming media IPv4 addresses for calls to the IP side from the remote IP device.
incoming port	Configure debug filtering for the incoming port.
outgoing media remote ipv4	Configure debug filtering for the outgoing media IPv4 addresses for calls to the IP side from the remote IP device.
outgoing port	Configure debug filtering for the outgoing port.
show call filter match-list	Display call filter match lists.

outgoing media remote ipv4

To configure debug filtering for the outgoing media remote IPv4 addresses for the voice gateway receiving the media stream, use the outgoing media remote ipv4 command in call filter match list configuration mode. To disable, use the **no** form of this command.

outgoing media remote ipv4 *ip_address*
no outgoing media remote ipv4 *ip_address*

Syntax Description

ip_address	IP address of the remote IP device

Command Default

No default behavior or values

Command Modes

Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example shows the voice call debug filter set to match outgoing media on the remote IP device, which has IP address 192.168.10.255:

```
call filter match-list 1 voice
 outgoing media remote ipv4 192.168.10.255
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
incoming media local ipv4	Configure debug filtering for the incoming media IPv4 addresses for calls to the IP side from the local voice gateway.
incoming media remote ipv4	Configure debug filtering for the incoming media IPv4 addresses for calls to the IP side from the remote IP device.
incoming port	Configure debug filtering for the incoming port.
outgoing media local ipv4	Configure debug filtering for the outgoing media IPv4 addresses for calls to the IP side from the local voice gateway
outgoing port	Configure debug filtering for the outgoing port.
show call filter match-list	Display call filter match lists.

outgoing port

To configure debug filtering for the outgoing port, use the outgoing port command in call filter match list configuration mode. To disable, use the **no** form of this command.

Cisco 2600, Cisco 3600, and Cisco 3700 Series
outgoing port {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-no*}
no outgoing port {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-no*}

Cisco 2600 and Cisco 3600 Series with a High-Density Analog Network Module (NM-HDA)
outgoing port {*slot-number*/*subunit-number*/*port*}
no outgoing port {*slot-number*/*subunit-number*/*port*}

Cisco AS5300
outgoing port *controller-number***:D**
no outgoing port *controller-number***:D**

Cisco AS5400
outgoing port *card*/*port***:D**
no outgoing port *card*/*port***:D**

Cisco AS5800
outgoing port {*shelf*/*slot*/*port***:D** | *shelf*/*slot*/*parent*:*port***:D**}
no outgoing port {*shelf*/*slot*/*port***:D** | *shelf*/*slot*/*parent*:*port***:D**}

Cisco MC3810
outgoing port *slot*/*port*
no outgoing port *slot*/*port*

Syntax Description

slot-number	Number of the slot in the router in which the VIC is installed. Valid entries are 0 to 3, depending on the slot in which it has been installed.
subunit-number	Subunit on the VIC in which the voice port is located. Valid entries are 0 or 1.
port	Voice port number. Valid entries are 0 and 1.
slot	The router location in which the voice port adapter is installed. Valid entries are 0 to 3.
port:	Indicates the voice interface card location. Valid entries are 0 and 3.
ds0-group-no	Indicates the defined DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

controller-number	T1 or E1 controller.
:D	D channel associated with ISDN PRI.

card	Specifies the T1 or E1 card. Valid entries for the *card* argument are 1 to 7.

port	Specifies the voice port number. Valid entries are 0 to 7.
:D	Indicates the D channel associated with ISDN PRI.

shelf	Specifies the T1 or E1 controller on the T1 card, or the T1 controller on the T3 card. Valid entries for the *shelf* argument are 0 to 9999.
slot	Specifies the T1 or E1 controller on the T1 card, or the T1 controller on the T3 card. Valid entries for the *slot* argument are 0 to 11.
port	Specifies the voice port number. • T1 or E1 controller on the T1 card --Valid entries are 0 to 11. • T1 controller on the T3 card--Valid entries are 1 to 28
:*port*	Specifies the value for the *parent* argument. The only valid entry is 0.
:D	Indicates the D channel associated with ISDN PRI.

slot	The slot argument specifies the number slot in the router in which the VIC is installed. The only valid entry is 1.
port	The port variable specifies the voice port number. Valid interface ranges are as follows: • T1--ANSI T1.403 (1989), Telcordia TR-54016. • E1-- ITU G.703. • Analog voice--Up to six ports (FXS, FXO, E & M). • Digital voice-- Single T1/E1 with cross-connect drop and insert, CAS and CCS signaling, PRI QSIG. • Ethernet--Single 10BASE-T. • Serial--Two five-in-one synchronous serial (ANSI EIA/TIA-530, EIA/TIA-232, EIA/TIA-449; ITU V.35, X.21, Bisync, Polled async).

Command Default

No default behavior or values

Command Modes

Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example shows the voice call debug filter set to match outgoing port 1/1/1 on a Cisco 3660 voice gateway:

```
call filter match-list 1 voice
 outgoing port 1/1/1
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
incoming port	Configure debug filtering for the incoming port.
show call filter match-list	Display call filter match lists.

outgoing signaling local ipv4

To configure debug filtering for the outgoing signaling local IPv4 addresses for the gatekeeper managing the signaling, use the outgoing signaling local ipv4 command in call filter match list configuration mode. To disable, use the **no** form of this command.

outgoing signaling local ipv4 *ip_address*
no outgoing signaling local ipv4 *ip_address*

Syntax Description

ip_address	IP address of the local voice gateway

Command Default

No default behavior or values

Command Modes

Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example shows the voice call debug filter set to match outgoing signaling on the local voice gateway, which has IP address 192.168.10.255:

```
call filter match-list 1 voice
 outgoing signaling local ipv4 192.168.10.255
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
incoming port	Configure debug filtering for the incoming port.
incoming signaling local ipv4	Configure debug filtering for the incoming signaling IPv4 addresses for calls to the IP side from the local voice gateway.
incoming signaling remote ipv4	Configure debug filtering for the incoming signaling IPv4 addresses for calls to the IP side from the remote IP device.
outgoing port	Configure debug filtering for the outgoing port.
outgoing signaling remote ipv4	Configure debug filtering for the outgoing signaling IPv4 addresses for calls to the IP side from the remote IP device.
show call filter match-list	Display call filter match lists.

outgoing signaling remote ipv4

To configure debug filtering for the outgoing signaling remote IPv4 addresses for the gatekeeper managing the signaling, use the outgoing signaling remote ipv4 command in call filter match list configuration mode. To disable, use the **no** form of this command.

outgoing signaling remote ipv4 *ip_address*
no outgoing signaling remote ipv4 *ip_address*

Syntax Description

ip_address	IP address of the remote IP device

Command Default

No default behavior or values

Command Modes

Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples

The following example shows the voice call debug filter set to match outgoing signaling on the remote IP device, which has IP address 192.168.10.255:

```
call filter match-list 1 voice
 outgoing signaling remote ipv4 192.168.10.255
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
incoming port	Configure debug filtering for the incoming port.
incoming signaling local ipv4	Configure debug filtering for the incoming signaling IPv4 addresses for calls to the IP side from the local voice gateway.
incoming signaling remote ipv4	Configure debug filtering for the incoming signaling IPv4 addresses for calls to the IP side from the remote IP device.
outgoing port	Configure debug filtering for the outgoing port.
outgoing signaling local ipv4	Configure debug filtering for the outgoing signaling IPv4 addresses for calls to the IP side from the local voice gateway.
show call filter match-list	Display call filter match lists.

output attenuation

To configure a specific output attenuation value or enable automatic gain control, use the **output attenuation**command in voice-port configuration mode. To disable the selected output attenuation value, use the **no** form of this command.

output attenuation {*decibels* | **auto-control** [*auto-dbm*]}
no output attenuation {*decibels* | **auto-control** [*auto-dbm*]}

Syntax Description

decibels	Attenuation, in decibels (dB), at the transmit side of the interface. Range is integers from -6 to 14. The default is 3.
auto-control	Enable automatic gain control.
auto-dbm	(Optional) Target speech level, in decibels per milliwatt (dBm), to be achieved at the transmit side of the interface. Range is integers from -30 to 3. The default is -9.

Command Default

For Foreign Exchange Office (FXO), Foreign Exchange Station (FXS), and ear and mouth (E&M) ports: *decibels*: 3 decibels*auto-dbm*: -9 dBm

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(1)MA	This command was implemented on the Cisco MC3810.
12.3(4)XD	The range of values for the *decibels* argument was increased.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	The **auto-control** keyword and *auto-dbm* argument were added.

Usage Guidelines

A system-wide loss plan must be implemented using both the **input gain** and **output attenuation** commands. You must consider other equipment (including PBXs) in the system when creating a loss plan. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there must be an attenuation of -6 dB between phones. Connections are implemented to provide -6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 3 dB.

You cannot increase the gain of a signal to the public switched telephone network (PSTN), but you can decrease it. If the voice level is too high, you can decrease the volume by either decreasing the input gain or increasing the output attenuation.

You can increase the gain of a signal coming into the router. If the voice level is too low, you can increase the input gain by using the **input gain** command.

The **auto-control**keyword and *auto-dbm* argument are available on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). The **auto-control**keyword enables automatic gain control, which is performed by the digital signal processor (DSP). Automatic gain control adjusts speech to a comfortable volume when it becomes too loud or too soft. Because of radio network loss and other environmental factors, the speech level arriving at a router from an LMR system could be very low. You can use automatic gain control to ensure that the speech is played back at a more comfortable level. Because the gain is inserted digitally, the background noise can also be amplified. Automatic gain control is implemented as follows:

- Output level: -9 dB
- Gain range: -12 dB to 20 dB
- Attack time (low to high): 30 milliseconds
- Attack time (high to low): 8 seconds

Examples

On the Cisco 3600 series router, the following example configures a 3-dB loss to be inserted at the transmit side of the interface:

```
voice-port 1/0/0
 output attenuation 3
```

On the Cisco 3600 series router, the following example configures a 3-dB gain to be inserted at the transmit side of the interface:

```
voice-port 1/0/0
 output attenuation -3
```

On the Cisco AS5300, the following example configures a 3-dB loss to be inserted at the transmit side of the interface:

```
voice-port 0:D
 output attenuation 3
```

Related Commands

Command	Description
comfort-noise	Generates background noise to fill silent gaps during calls if VAD is activated.
echo-cancel enable	Enables the cancellation of voice that is sent out the interface and received back on the same interface.
input gain	Configures a specific input gain value or enables automatic gain control for a voice port.

overhead

To configure the overhead negotiated bandwidth percentage, use the **overhead** command in media profile configuration mode. To disable the configuration, use the **no** form of the command.

overhead {**audio** | **video**} *percentage*
no overhead {**audio** | **video**}

Syntax Description

audio	Configures the audio overhead percentage.
video	Configures the video overhead percentage.
percentage	Overhead percentage. The range is from 0 to 50.

Command Default

Overhead negotiated bandwidth is not configured.

Command Modes

Media profile configuration (cfg-mediaprofile)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

The overhead bandwidth is the extra bandwidth apart from the negotiated bandwidth for audio and video calls. Hence, the total policing bandwidth is:

Policing bandwidth = negotiated bandwidth + (1 + % overhead bandwidth)

Examples

The following example shows how to configure an overhead bandwidth of 10 percent for audio codecs and 20 percent for video codecs:

```
Router> enable
Router# configure terminal
Router(config)# media profile police 1
Router(cfg-mediaprofile)# overhead audio 10
Router(cfg-mediaprofile)# overhead video 20
```

Related Commands

Command	Description
media profile police	Configures the media policing profile.