# N

# name (dial peer cor custom)

To specify the name for a custom class of restrictions (COR), use the **name** command in dial peer COR custom configuration mode. To remove a specified COR, use the **no** form of this command.

**name** *class-name*
**no name** *class-name*

## Syntax Description

| | |
|---|---|
| *class-name* | Name that describes the specific COR. |

## Command Default

No default behavior or values.

## Command Modes

Dial peer COR custom configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(3)T | This command was introduced. |

## Usage Guidelines

The **dial-peer cor custom** and **name** commands define the names of capabilities on which to apply COR operation. Examples of names might include any of the following: call1900, call527, call9, or call 911. You must define the capabilities before you specify the COR rules.

You can define a maximum of 64 COR names.

## Examples

The following example defines three COR names:

```
dial-peer cor custom
 name 900_call
 name 800_call
 name catchall
```

## Related Commands

| Command | Description |
|---|---|
| **dial-peer cor custom** | Specifies that named CORs apply to dial peers. |
| **name** | Assigns a name to the internal adapter. |

# nat (sip-ua)

To use the SIP Network Address Translation (NAT) global configuration, use the **nat** command in SIP user agent configuration mode. To disable the **nat** configuration, use the **no** or **default** form of this command.

**nat auto** { **force-on** | **force-off** }
**no nat**

| | |
|---|---|
| **auto** | Sets the symmetric NAT endpoint role to auto. Autodetect subscriber in a remote subnet when located behind a NAT. |
| **force-on** | Sets the symmetric NAT endpoint role to force-on. Assume that all remote subscribers are behind the NAT device. |

## Command Modes

SIP user agent configuration (sip-ua)

Voice class tenant configuration (config-class)

Voice service SIP configuration (conf-serv-sip)

| Release | Modification |
|---|---|
| 12.2(13)T | This command was introduced. |
| Cisco IOS XE Dublin 17.10.1a | Introduced support for YANG models. |

## Examples

The following example shows how to set the endpoint role in connection setup to active:

```
Router(config)# sip-ua
Router(config-sip-ua)# nat auto

Router(config)# sip-ua
Router(config-sip-ua)# nat force-on
```

## Related Commands

| Command | Description |
|---|---|
| **nat symmetric check-media-src** | Enables source media checking for symmetric NAT. |

# nat media-keepalive

To enable media keepalive packets transmission for the specified interval of time (in seconds) at tenant or global level, use the **nat media-keepalive** command in voice class tenant configuration (config-class) or voice service SIP configuration (conf-serv-sip) mode. To disable the **nat** configuration, use the **no** or **default** form of this command.

**nat** { **auto** | **force-on** | **media-keepalive** [*interval*] }
**no nat**
**default nat**

## Syntax Description

| | |
|---|---|
| **media-keepalive** | Specifies media keepalive to subscriber if it's located behind NAT. |
| *interval* | Specifies keepalive interval configured in seconds. Range is 1—50. Default is 10. |

## Command Default

If no value is specified, default interval value is set to 10.

## Command Modes

Voice class tenant configuration (config-class)

Voice service SIP configuration (conf-serv-sip)

## Command History

| Release | Modification |
|---|---|
| Cisco IOS XE 17.13.1a<br>Cisco IOS XE Dublin 17.12.2 | This command was introduced. |

## Examples

The following example shows how to configure media keepalive at global level:

```
Device(config)# voice service voip
Device(config-voi-serv)# sip
Device(config-serv-sip)# nat media-keepalive 20
```

The following example shows how to configure media keepalive at tenant level:

```
Device(config)# voice class tenant 1
Device(config-class)# nat media-keepalive 35
```

# nat symmetric check-media-src

To enable the gateway, to check the media source of incoming Real-time Transport Protocol (RTP) packets in symmetric Network Address Translation (NAT) environments, use the **nat symmetric check-media-src** command in SIP user agent configuration mode. To disable media source checking, use the **no** form of this command.

**nat symmetric check-media-src**
**no nat symmetric check-media-src**

## Syntax Description

This command has no arguments or keywords.

## Command Default

Media source checking is disabled.

## Command Modes

SIP user agent configuration (sip-ua)

## Command History

| Release | Modification |
|---|---|
| 12.2(13)T | This command was introduced. |

## Usage Guidelines

This command provides the ability to enable or disable symmetric NAT settings for the Session Initiation Protocol (SIP) user agent. Use the **nat symmetric check-media-src command to configure the gateway to check the media source address and port of the first incoming**RTP **packet. Checking for media packets is automatically enabled if the gateway receives the direction role "active or both".**

## Examples

The following example enables checking the media source:

```
Router(config)# sip-ua
Router(config-sip-ua)# nat symmetric check-media-src
```

## Related Commands

| Command | Description |
|---|---|
| **nat symmetric role** | Defines endpoint settings to initiate or accept a connection for symmetric. |

# nat symmetric role

To define endpoint settings to initiate or accept a connection for symmetric Network Address Translation (NAT) configuration, use the **nat symmetric role** command in SIP user agent configuration mode. To disable the **nat symmetric role**configuration, use the **no** form of this command.

**nat symmetric role** {**active** | **passive**}
**no nat symmetric role** {**active** | **passive**}

## Syntax Description

| | |
|---|---|
| **active** | Sets the symmetric NAT endpoint role to active, originating an outgoing connection. |
| **passive** | Sets the symmetric NAT endpoint role to passive, accepting an incoming connection to the port number on the m=line of the Session Description Protocol (SDP) body sent from the SDP body to the other endpoint. |

## Command Default

The endpoint settings to initiate or accept connections for NAT configuration are not defined..

## Command Modes

SIP user agent configuration (sip-ua)

## Command History

| Release | Modification |
|---|---|
| 12.2(13)T | This command was introduced. |

## Usage Guidelines

**This command provides the ability to specify symmetric NAT endpoint settings for the SIP user agent. If the gateway does not receive the direction role, use the nat symmetric role command to define endpoint settings to initiate or accept a connection for symmetric NAT configuration. This is achieved by setting the symmetric NAT endpoint role to active or passive , respectively. Cisco recommends that you use the nat symmetric role command under the following conditions:**

- Endpoints are aware of their presence inside or outside of NAT
- Endpoints parse and process direction:<role> in SDP

If the endpoints conditions are not satisfied, you may not achieve the desired results when you configure the **nat symmetric role command.**

## Examples

The following example shows how to set the endpoint role in connection setup to active:

```
Router(config)# sip-ua
Router(config-sip-ua)# nat symmetric role active
```

## Related Commands

| Command | Description |
|---|---|
| **nat symmetric check-media-src** | Enables source media checking for symmetric NAT. |

# neighbor (annex g)

To configure the neighboring border elements (BEs) that interact with the local BE for the purpose of obtaining addressing information and aiding in address resolution, enter the **neighbor** command in Annex G configuration mode. To reset the default value, use the no form of this command.

**neighbor** *ip-address*
**no neighbor**

**Syntax Description**

| | |
|---|---|
| *ip -address* | IP address of the neighbor that is used for exchanging Annex G messages. |

**Command Default**

No default behavior or values

**Command Modes**

Annex G configuration

**Command History**

| Release | Modification |
|---|---|
| 12.2(2)XA | This command was introduced. |
| 12.2(4)T | This command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release. |
| 12.2(2)XB1 | This command was implemented on the Cisco AS5850. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release. |

**Examples**

The following example configures a neighboring BE that has an IP address and border element ID:

```
Router(config)# call-router h323-annexg be20
Router(config-annexg)# neighbor 121.90.10.42
Router(config-annexg-neigh)# id be30
Router(config-annexg-neigh)# exit
```

**Related Commands**

| Command | Description |
|---|---|
| **advertise** | Controls the types of descriptors that the BE advertises to its neighbors. |
| **call -router** | Enables the Annex G border element configuration commands. |
| **id** | Configures the local ID for the neighboring BE. |
| **port** | Configures the port number of the neighbor that is used for exchanging Annex G messages. |
| **query -interval** | Configures the interval at which the local BE will query the neighboring BE. |

# neighbor (tgrep)

To create a TGREP session with another device, use the neighbor command in TGREP configuration mode. To disable a TRIP connection, use the **no** form of this command.

**neighbor ip_address**
**no neighbor ip_address**

## Syntax Description

| | |
|---|---|
| *ip_address* | IP address of a peer device with which TGREP information will be exchanged. |

## Command Default

No neighboring devices are defined

## Command Modes

TGREP configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(1) | This command was introduced. |

## Examples

The following example shows that the gateway with the IP address 192.116.56.10 is defined as a neighbor for ITAD 1234:

```
Router(config)# tgrep local-itad 1234
Router(config-tgrep)# neighbor 192.116.56.10
```

## Related Commands

| Command | Description |
|---|---|
| tgrep local - itad | Enters TGREP configuration mode and defines an ITAD. |

# network-clock base-rate

To configure the network clock base rate for universal I/O serial ports 0 and 1, use the **network-clock base-rate** command in global configuration mode. To disable the current network clock base rate, use the no form of this command.

**network-clock base-rate** {**56k** | **64k**}
**no network-clock base-rate** {**56k** | **64k**}

## Syntax Description

| | |
|---|---|
| **56k** | Sets the network clock base rate to 56 kbps. |
| **64k** | Sets the network clock base rate to 64 kbps. |

## Command Default

56 kbps

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)MA | This command was introduced on the Cisco MC3810. |

## Usage Guidelines

This command applies to Voice over Frame Relay and Voice over ATM.

## Examples

The following example sets the network clock base rate to 64 kbps:

```
network-clock base-rate 64k
```

## Related Commands

| Command | Description |
|---|---|
| **network -clock-select** | Uses the network clock source to provide timing to the system backplane PCM bus. |
| **network -clock-switch** | Configures the switch delay time to the next priority network clock source when the current network clock source fails. |

# network-clock-participate

To allow the ports on a specified network module or voice/WAN interface card (VWIC) to use the network clock for timing, use the **network-clock-participate** command in global configuration mode. To restrict the device to use only its own clock signals, use the **no** form of this command.

**network-clock-participate** [{**slot** *slot-number* | **wic** *wic-slot* | **aim** *aim-slot-number*}]
**no network-clock-participate** [{**nm** *slot* | **wic** *wic-slot*}]

## Syntax Description

| | |
|---|---|
| **slot** *slot -number* | (Optional) Network module slot number on the router chassis. Valid values are from 1 to 6. |
| **wic** *wic -slot* | Configures the WAN interface card (WIC) slot number on the router chassis. Valid values are 0 or 1. |
| **aim** *aim -slot-number* | Configures the Advanced Integration Module (AIM) in the specified slot. The aim-slot-number values are 0 or 1 for the Cisco 3660 and 0 or 1 for the Cisco 3725, and Cisco 3745. |

## Command Default

No network clocking is enabled, and interfaces are restricted to using the clocking generated on their own modules.

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(5)XM | This command was introduced on the Cisco 3660. |
| 12.2(4)T | This command was integrated into Cisco IOS Release 12.2(4)T. |
| 12.2(2)XB | The **slot** keyword was replaced by the **nm** keyword and the **wic** keyword and the *wic-slot*argument were added. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. |
| 12.2(15)T | This command was integrated into Cisco IOS Release 12.2(15)T with support for the Cisco 3660, Cisco 3725, and Cisco 3745. Clocks can be synchronized on two ports. The **aim** keyword was added. The **nm** keyword was replaced by the **slot** keyword. |
| 12.4(15)T9 | This command was integrated into Cisco IOS Release 12.4(15)T9, and support was added for the NM-CEM-4SER modules. |

## Usage Guidelines

This command is used for ATM segmentation and reassembly or digital signal processing and Cisco 3660, Cisco 3725, and Cisco 3745 routers.

This command applies to any network module with T1/E1 controllers to provide clocks from a central source (MIX module for the Cisco 3660) to the network module and to the port on the network module. Then that port can be selected as the clock source with the **network-clock-select** command to supply clock to other ports or network modules that choose to participate in network clocking with the **network-clock-participate** command. This command synchronizes the clocks for two ports.

On the Cisco 3700 series, you must use the **network-clock-participate** command and either the **wic** *wic-slot*keyword and argument or the **slot** *slot-number* keyword and argument.

**Note** If the AIM takes its clock signals from a T1 or E1 controller, it is mandatory to use the **network-clock-select** and **network-clock-participate** commands for ATM. The clocks for the ATM and voice interfaces do not need to be synchronous, but improved voice quality may result if they are.

**Note** The only VWICs that can participate in network clocking are digital T1/E1 packet voice trunk network modules (NM-HDV), and Fast Ethernet network modules (NM-2W, NM-1FE. and NM-2FE).

**Note** Beginning with Cisco IOS Release 12.4(15)T9, the **network-clock-participate** command can also be used for the NM-CEM-4SER modules. When the **network-clock-participate** command is configured, the clock is derived from the backplane. When the **no network-clock-participate** command is configured, the local oscillator clock is used.

## Examples

The following example configures the network module in slot 5 to participate in network clocking on a Cisco 3660 with a MIX module:

```
network-clock-participate slot 5
network-clock-select 1 e1
```

The following example on a Cisco 3700 series router specifies that the AIM participates in network clocking and selects port E1 0/1 to provide the clock signals.

```
Router(config)# network-clock-participate wic 0
Router(config)# network-clock-participate aim 0
Router(config)# network-clock-select 2 E1 0/1
```

The following example on a Cisco 3660 specifies the slot number that participates in network clocking and selects port E1 5/0:

```
Router(config)# network-clock-participate slot 5
Router(config)# network-clock-select 1 E1 5/0
```

## Related Commands

| Command | Description |
|---|---|
| **network-clock-select** | Specifies selection priority for the clock sources. |
| **network-clock-source** | Selects the port to be the clock source to supply clock resources to other ports or network modules. |

# network-clock select

To name a source to provide timing for the network clock and to specify the selection priority for this clock source, use the **network-clock select** command in global configuration mode. To cancel the network clock selection, use the **no** form of this command.

**Cisco ASR 1000 Series**

**network-clock select** {*priority* [{**bits** [{**R0** | **R1**}] {**e1** [{**crc4** | **no-crc4** | **unframed**}] | **t1** [{**esf** | **sf** | **unframed**}]} | **controller** *type number* | **global** | **interface** *type number* | **local** | **system**}] | **option** {**1** | **2**}}

**no network-clock select** *priority* [{**global** | **local**}]

**Cisco 7600 Series and Cisco 10000 Series**

**network-clock select** *priority* {**controller** *type number* | **interface** *type number* | **slot** *number* | **system**} [{**global** | **local**}]

**no network-clock select** *priority* [{**global** | **local**}]

## Syntax Description

| | |
|---|---|
| *priority* | Selection priority for the clock source (1 is the highest priority). The range is 1 to 6.<br>The clock with the highest priority is selected to drive the system time division multiplexing (TDM) clocks. When the higher-priority clock source fails, the next-higher-priority clock source is selected. |
| **bits** | (Optional) Derives network timing from the central office (CO) Building Integrated Timing Supply (BITS) clock. |
| **R0** | (Optional) Specifies Route Processor 0 BITS as the source slot. |
| **R1** | (Optional) Specifies Route Processor 1 BITS as the source slot. |
| **e1** | (Optional) Configures the BITS interface to use an E1 connection. |
| **crc4** | (Optional) Configures the E1 BITS interface framing with Cyclic Redundancy Check 4 (CRC4). |
| **no-crc4** | (Optional) Configures the E1 BITS interface framing with no CRC4. |
| **unframed** | (Optional) Configures the BITS interface with clear channel. |
| **t1** | (Optional) Configures the BITS interface to use a T1 connection. |
| **esf** | (Optional) Configures the T1 BITS interface with the Extended Super Frame (ESF) framing standard. |
| **sf** | (Optional) Configures the T1 BITS interface with the Super Frame (SF) framing standard. |
| **controller** *type number* | Specifies the controller to be the clock source. |
| **interface** *type number* | Specifies the interface to be the clock source. |

| | |
|---|---|
| **slot** *number* | Specifies the slot to be the clock source. The range is 1 to 6. |
| **global** | (Optional) Configures the source as global. |
| **local** | (Optional) Configures the source as local. |
| **system** | Specifies the system clock as the clock source. |
| **option** | Specifies the standards for the network option. The applicable values are as follows:<br>• 1—Network option I is the ITU G-813 standard.<br>• 2—Network option II (Gen1) is the Bellcore GR-1244/GR-253 (stratum 3) and ITU G-813 standard. This is the default value.<br>**Note** The network options are available only in the RP2 platform. |

## Command Default

The router uses the system clock (also called free-running mode).

**Note** Because default clock values are derived from an external source, they can fall outside the configurable range.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 11.3 MA | This command was introduced on the Cisco MC3810. |
| 12.0(3)XG | The BVM as a possible network clock source was added. |
| 12.1(5)XM | This command was implemented on the Cisco 3660. The keywords **t1** and **e1** were introduced. |
| 12.2(4)T | This command was integrated into Cisco IOS Release 12.2(4)T. |
| 12.2(2)XB | This command was implemented on the Cisco 2600 series and Cisco 3660 with AIMs installed. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. |
| 12.2(15)T | This command was implemented on the Cisco 2600XM, Cisco 2691, Cisco 3725, and Cisco 3745. |
| 12.3(8)T4 | This command was integrated into Cisco IOS Release 12.3(8)T4 and the **bri**keyword was added. Support was also added for the Cisco 2800 series. |
| 12.3(11)T | This command was integrated into Cisco IOS Release 12.3(11)T and the **atm**keyword was added. Support was also added for the Cisco 3800 series. |
| Cisco IOS XE Release 2.1 | This command was introduced in a release earlier than Cisco IOS Release 2.1. |
| 15.0(1)S | This command was integrated into a release earlier than Cisco IOS Release 15.0(1)S. |

| Release | Modification |
|---|---|
| Cisco IOS XE Release 3.1 | This command was modified. This command was implemented on the Cisco ASR 1000 platform. The **option** keyword was added. |

## Usage Guidelines

**When an active clock source fails, the system chooses the next-lower-priority clock source that is specified by this command. When a higher-priority clock source becomes available, the system automatically reselects it.**

You can specify up to five clock priorities. The highest-priority active interface in the router supplies the primary reference source to all other interfaces that require network clock synchronization services.

For timing sources, the Route Processor can receive timing information through its BITS interface or through a TDM-based Shared Port Adapter (SPA). For some telecommunications deployments, BITS clocking is required to provide global clocking synchronization of network equipment in the end-to-end data path. A BITS clock can be supplied to the network clock module using a T1 or E1 connection.

If a controller is specified in the clock source hierarchy, you must configure that controller for line timing (by using the appropriate **clock source line** command for the controller). Any controller that is not currently acting as the clock source will automatically operate in loop timing mode. Both controllers can be given different clock source priority values. For more information, see the Cisco IOS Interface and Hardware Component Command Reference .

**Note** To minimize backplane clock shifts, the **no network-clock select** command does not take effect until you return to EXEC mode by entering **exit** or **end**. This process minimizes the number of times that clock sources are configured.

Use the **show network-clocks** command to display clock priorities that are configured on the router.

## Examples

The following example shows how to configure the network clock as revertive and assign clock sources to two priorities:

```
Router> enable
Router# configure terminal
Router(config)# network-clock revertive
Router(config)# network-clock select 1 bits R0 e1
Router(config)# network-clock select 2 interface GigabitEthernet 0/0/1
```

The following example shows how to configure the network option for network clock.

Router(config)# **network-clock select option 1**

## Related Commands

| Command | Description |
|---|---|
| **network-clock-participate** | Configures a network module to participate in network clocking. |
| **network-clock-switch** | Configures the switch delay time to the next-priority network clock source when the current network clock source fails or a higher-priority clock source is up and available. |
| **show network-clocks** | Displays the network clock configuration and current primary clock source. |

# network-clock-switch

To configure the switch delay time to the next priority network clock source when the current network clock source fails, use the **network-clock-switch** command in global configuration mode. To cancel the network clock delay time selection, use the no form of this command.

**network-clock-switch** [{*switch-delay* | **never**}] [{*restore-delay* | **never**}]
**no network-clock-switch**

## Syntax Description

| | |
|---|---|
| *switch -delay* | (Optional) Delay time, in seconds, before the next-priority network clock source is used when the current network clock source fails. Range is from 0 to 99. Default is 10. |
| **never** | (Optional) No delay time before the current network clock source recovers. |
| *restore -delay* | (Optional) Delay time, in seconds, before the current network clock source recovers. Range is from 0 to 99. |
| **never** | (Optional) No delay time before the next-priority network clock source is used when the current network clock source fails. |

## Command Default

10 seconds

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)MA | This command was introduced on the Cisco MC3810. |

## Usage Guidelines

This command applies to Voice over Frame Relay and Voice over ATM.

## Examples

The following example switches the network clock source after 20 seconds and sets the delay time before the current network clock source recovers to 20 seconds:

```
network-clock-switch 20 20
```

## Related Commands

| Command | Description |
|---|---|
| **network -clock-select** | Uses the network clock source to provide timing to the system backplane PCM bus. |

# noisefloor

To configure the noise level, in dBm, above which noise reduction (NR) will operate, use the **noisefloor** command in media profile configuration mode. To disable the configuration, use the **no** form of this command.

**noisefloor** *level*
**no noisefloor** *level*

### Syntax Description

| | |
|---|---|
| *level* | Minimum noise level in dBm. The range is from -58 to -20. |

### Command Default

The default value is -48 dBm.

### Command Modes

Media profile configuration (cfg-mediaprofile)

### Command History

| Release | Modification |
|---|---|
| 15.2(2)T | This command was introduced. |
| 15.2(3)T | This command was modified. Support for the Cisco Unified Border Element (Cisco UBE) was added. |

### Usage Guidelines

Use the **noisefloor** command to configure the noise level, in dBm, above which noise reduction (NR) will operate. NR will allow noises quieter than this level to pass without processing. You must create a media profile for noise reduction and then configure the noise level. Signal levels start at 0 dBm (extremely loud) and quieter levels are more negative. The default value of -48 dBm is very quiet.

### Examples

The following example shows how to create a media profile to configure noise reduction parameters:

```
Device> enable
Device# configure terminal
Device(config)# media profile nr 200
Device(cfg-mediaprofile)# noisefloor -50
Device(cfg-mediaprofile)# end
```

### Related Commands

| Command | Description |
|---|---|
| **intensity** | The intensity or depth of the noise reduction process. |
| **media profile nr** | Creates a media profile to configure noise reduction parameters. |

# non-linear

To enable nonlinear processing (NLP) in the echo canceller and set its threshold or comfort-noise attenuation, use the **non-linear** command in voice-port configuration mode. To disable nonlinear processing, use the **no** form of this command.

**non-linear** [{**comfort-noise attenuation** {**0db** | **3db** | **6db** | **9db**} | **threshold** *dB*}]
**no non-linear** [{**comfort-noise attenuation** | **threshold**}]

## Syntax Description

| | |
|---|---|
| **0db** \| **3db** \| **6db** \| **9db** | (Optional) Attenuation level of the comfort noise in dB. Default is **0db**, which means that comfort noise is not attenuated. |
| **threshold** *dB* | (Optional) Sets the threshold in dB. Range is -15 to -45. Default is -21. **Note** This keyword is not supported when using the extended G.168 echo canceller. |

## Command Default

NLP is enabled; comfort-noise attenuation is disabled; threshold is -21 dB.

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced. |
| 12.2(11)T | The **threshold** keyword was added. |
| 12.2(13)T | This command was implemented on routers that support the extended G.168 echo canceller. |
| 12.3(6) | The **comfort-noise** keyword was added. |
| 12.4 | The default setting for comfort-noise attenuation was changed from 0db to 6db. |

## Usage Guidelines

This command enables functionality that is also generally known as residual echo suppression. Use this command to shut off any signal if no near-end speech is detected. Enabling this command normally improves performance, although some users might perceive truncation of consonants at the end of sentences when this command is enabled.

Use the **comfort-noise** keyword if the comfort noise generated by the NLP sounds like hissing. Using this keyword makes the hissing sound less audible. The default setting for comfort-noise attenuation is 6db to achieve the highest satisfaction in voice quality.

**Note** The **echo-cancel enable** command must be enabled for this command to take effect.

## Examples

The following example enables nonlinear call processing on a Cisco 3600 series router:

```
voice-port 1/0/0
 non-linear
```

The following example sets the attenuation level to 9 dB on a Cisco 3600 series router:

```
voice-port 1/0/0
 non-linear comfort-noise attenuation 9db
```

## Related Commands

| Command | Description |
| --- | --- |
| **echo -cancel enable** | Enables echo cancellation for voice that is sent and received on the same interface. |

# notify (MGCP profile)

To specify the order in which automatic number identification (ANI) and dialed number identification service (DNIS) digits are reported to the Media Gateway Control Protocol (MGCP) call agent, use the **notify**command in MGCP profile configuration mode. To revert to the default, use the **no** form of this command.

**notify** {**ani-dnis** | **dnis-ani**}
**no notify** {**ani-dnis** | **dnis-ani**}

## Syntax Description

| | |
|---|---|
| **ani-dnis** | ANI digits are sent in the first notify message, followed by DNIS. This is the default. |
| **dnis-ani** | DNIS digits are sent in the first notify message, followed by ANI. |

## Command Default

The default order is ANI first and DNIS second.

## Command Modes

MGCP profile configuration

## Command History

| Release | Modification |
|---|---|
| 12.4(4)T | This command was introduced. |

## Usage Guidelines

This command controls the order of ANI and DNIS when using the Feature Group D (FGD) Exchange Access North American (EANA) protocol on a T1 interface. Selecting the **ani-dnis** keyword causes the ANI digits to be sent in the first NTFY message to the MGCP call agent and the DNIS digits to be sent in a second NTFY message. Selecting the **dnis-ani** keyword causes the DNIS digits to be sent in the first NTFY message to the MGCP call agent and the ANI digits to be sent in a second NTFY message.

## Examples

The following example sets the digit order to DNIS first and ANI second for the default MGCP profile:

```
Router(config)# mgcp profile default
Router(config-mgcp-profile)# notify dnis-ani
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp package-capability** | Specifies an MGCP package capability type for a media gateway. |
| **mgcp profile** | Defines an MGCP profile to be associated with one or more MGCP endpoints |
| **show mgcp** | Displays MGCP configuration information. |
| **show mgcp profile** | Displays information for MGCP profiles. |

# notify redirect

To enable application handling of redirect requests for all VoIP dial peers on a Cisco IOS voice gateway, use the **notify redirect**command in voice service VoIP configuration mode. To disable application handling of redirect requests on the gateway, use the **no** form of this command. To return the gateway to the default **notify redirect** command settings, use the **default** form of this command.

**notify redirect** {**ip2ip** | **ip2pots**}
**no notify redirect** {**ip2ip** | **ip2pots**}
**default notify redirect** {**ip2ip** | **ip2pots**}

## Syntax Description

| | |
|---|---|
| **ip2ip** | Enables notify redirection for IP-to-IP calls. |
| **ip2pots** | Enables notify redirection for IP-to-IP calls for IP-to-POTS calls. |

## Command Default

Notify redirection for IP-to-IP calls is enabled.

Notify redirection for IP-to-POTS calls is disabled.

Notify redirection for Session Initiation Protocol (SIP) phones registered to Cisco Unified Communications Manager Express (Cisco Unified CME) is enabled.

## Command Modes

Voice service VoIP configuration (conf-voi-serv)

## Command History

| Release | Modification |
|---|---|
| 12.4(4)T | This command was introduced. |
| 15.1(1)T | This command was integrated into Cisco IOS Release 15.1(1)T. The following default behavior was added: Notify redirection for SIP phones registered to Cisco Unified CME is enabled. |
| Cisco IOS XE Cupertino 17.7.1a | Introduced support for YANG models. |

## Usage Guidelines

Use this command to enable notify redirection globally on a gateway. Use the **notify redirect** command in dial peer voice configuration mode to configure notify redirection settings for IP-to-IPand IP-to-POTS calls on a specific inbound dial peer on a gateway.

**Note** This command is supported on Cisco Unified Communications Manager Express (Cisco Unified CME), release 3.4 and later releases and on Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) release 3.4 and later releases. However, to use the **notify redirect** command in voice service VoIP configuration mode on compatible Cisco Unified SIP SRST devices, you must first use the **allow-connections** command to enable the corresponding call flows on the SRST gateway.

## Examples

The following is partial sample output from the **show running-config** command showing that notify redirection has been set up globally for both IP-to-IP and IP-to-POTS calling (because support of IP-to-IP calls is enabled by default, the ip2ip setting does not appear in the output).

```
voice service voip
 notify redirect ip2pots
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to sip
 no supplementary-service h450.2
 no supplementary-service h450.3
 sip
 registrar server expires max 600 min 60
```

## Related Commands

| Command | Description |
|---|---|
| **allow-connections** | Allows connections between specific endpoint types in a VoIP network. |
| **notify redirect (dial peer)** | Enables application handling of redirect requests on a specific VoIP dial peer on a Cisco IOS voice gateway. |

# notify redirect (dial peer)

To enable application handling of redirect requests on a specific VoIP dial peer on a Cisco IOS voice gateway, use the **notify redirect** command in dial peer voice configuration mode. To disable notify redirection on the gateway, use the **no** form of this command. To return the gateway to the default notify redirection settings, use the **default** form of this command.

**notify redirect** {**ip2ip** | **ip2pots**}
**no notify redirect** {**ip2ip** | **ip2pots**}
**default notify redirect** {**ip2ip** | **ip2pots**}

## Syntax Description

| | |
|---|---|
| **ip2ip** | Specifies that the notify redirect command is applied to IP-to-IP calls. |
| **ip2pots** | Specifies that the notify redirect command is applied to IP-to-POTS calls. |

## Command Default

Notify redirection for IP-to-IP is enabled. Notify redirection for IP-to-POTS is disabled.

Notify redirection for Session Initiation Protocol (SIP) phones registered to Cisco Unified Communications Manager Express (Cisco Unified CME) is enabled.

## Command Modes

Dial peer voice configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 12.4(4)T | This command was introduced. |
| 15.1(1)T | This command was integrated into Cisco IOS Release 15.1(1)T. The following default behavior was added: Notify redirection for SIP phones registered to Cisco Unified CME is enabled. |

## Usage Guidelines

Use this command in dial peer configuration mode to configure IP-to-IP and IP-to-POTS calls on an inbound dial peer on a Cisco IOS voice gateway. This command configures notify redirection settings on a per-dial-peer basis.

When notify redirect is enabled in dial peer voice configuration mode, the configuration for the specific dial peer is activated only if the dial peer is an inbound dial peer. To enable notify redirect globally on a Cisco IOS voice gateway, use the **notify redirect** command in voice service VoIP configuration mode.

**Note** This command is supported on Cisco Unified Communications Manager Express (Cisco Unified CME), release 3.4 and later releases and Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) release 3.4 and later releases. However, to use the **notify redirect** command in voice service VoIP configuration mode on compatible Cisco Unified SIP SRST devices, you must first use the **allow-connections** command to enable the corresponding call flows on the SRST gateway.

## Examples

The following is partial sample output from the **show running-config** command showing that notify redirection is enabled for both IP-to-IP and IP-to-POTS calls on VoIP dial peer 8000 (because support of IP-to-IP calls is enabled by default, the ip2ip setting does not appear in the output):

```
dial-peer voice 8000 voip
 destination-pattern 80..
 notify redirect ip2pots
 session protocol sipv2
 session target ipv4:209.165.201.15
 dtmf-relay rtp-nte
 codec g711ulaw
!
```

## Related Commands

| Command | Description |
| --- | --- |
| **allow-connections** | Allows connections between specific endpoint types in a VoIP network. |
| **notify redirect** | Enables application handling of redirect requests for all VoIP dial peers on a Cisco IOS voice gateway. |

# notify telephone-event

To configure the maximum interval between two consecutive NOTIFY messages for a particular telephone event, use the **notify telephone-event** command in SIP UA configuration mode or voice class tenant configuration mode. To reset the interval to the default value, use the **no** form of this command.

**notify telephone-event max-duration** *milliseconds* **[system]**
**no notify telephone-event**

## Syntax Description

| | |
|---|---|
| **max-duration** *milliseconds* | Time interval between consecutive NOTIFY messages for a single DTMF event, in milliseconds. Range is from 40 to 3000. Default is 2000. |
| **system** | Specifies that the NOTIFY messages for a particular telephone event 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

2000 milliseconds

## Command Modes

SIP UA configuration (config-sip-ua)

Voice class tenant configuration (config-class)

## Command History

| Release | Modification |
|---|---|
| 12.2(15)ZJ | This command was introduced. |
| 12.3(4)T | This command was integrated into Cisco IOS Release 12.3(4)T. |
| 15.0(1)M | This command was modified. The acceptable value range for the *milliseconds* argument was expanded (the lower end of the range was changed from 500 to 40). |
| 12.4(24)T3 | This command was modified. The acceptable value range for the *milliseconds* argument was expanded (the lower end of the range was changed from 500 to 40). |
| 15.6(2)T and IOS XE Denali 16.3.1 | This command was modified to include the keyword: **system**. |
| Cisco IOS XE Dublin 17.10.1a | Introduced support for YANG models. |

## Usage Guidelines

The **notify telephone-event** command works with the **dtmf-relay sip-notify** command. The **dtmf-relay sip-notify** command forwards out-of-band DTMF tones by using SIP NOTIFY messages. The **notify telephone-event** command sets the maximum time interval between consecutive NOTIFY messages for a single DTMF event. The maximum time is negotiated between two SIP endpoints and the lowest duration value is the one selected. This duration is negotiated during call establishment as part of negotiating the SIP-NOTIFY DTMF relay.

The originating gateway sends an indication of DTMF relay in an Invite message using the SIP Call-Info header. The terminating gateway acknowledges the message with an 18x/200 Response message, also using the Call-Info header. The set duration appears in the Call-Info header in the following way:

```
Call-Info: <sip: address>; method="Notify;Event=telephone-event;Duration=msec"
```

For example, if the maximum duration of gateway A is set to 1000 ms, and gateway B is set to 700 ms, the resulting negotiated duration would be 700 ms. Both A and B would use the value 700 in all of their NOTIFY messages for DTMF events.

## Examples

The following example sets the maximum duration for a DTMF event to 40 ms.

```
Router(config)# sip-ua
Router(config-sip-ua)# notify telephone-event max-duration 40
```

The following example sets the maximum duration for a DTMF event in the voice class tenant configuration mode:

```
Router(config-class)# notify telephone-event max-duration system
```

## Related Commands

| Command | Description |
|---|---|
| dtmf-relay sip-notify | Forwards DTMF tones using SIP NOTIFY messages. |

# notify ignore substate

To ignore the Subscription-State header, use the **notify ignore substae** command in SIP UA configuration mode or voice class tenant configuration mode. To reset the interval to the default value, use the **no** form of this command.

**notify ignore substate**
**no notify ignore substate**

## Command Modes

SIP UA configuration (config-sip-ua)

Voice class tenant configuration (config-class)

## Command History

| Release | Modification |
|---|---|
| 12.2(15)ZJ | This command was introduced. |
| Cisco IOS XE Dublin 17.10.1a | Introduced support for YANG models. |

## Examples

The following is an example:

```
Router(config)# sip-ua
Router(config-sip-ua)# notify ignore substate
```

# nsap

To specify the network service access point (NSAP) address for a local video dial peer, use the **nsap**command in dial-peer configuration mode. To remove any configured NSAP address from the dial peer, use the **no** form of this command.

**nsap** *nsap-address*
**no nsap**

## Syntax Description

| | |
|---|---|
| *nsap -address* | A 40-digit hexadecimal number; the number must be unique on the device. |

## Command Default

No NSAP address for a video dial peer is configured

## Command Modes

Dial-peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(5)XK | This command was introduced for ATM video dial-peer configuration on the Cisco MC3810. |
| 12.0(7)T | This command was integrated into Cisco IOS Release 12.0(9)T. |

## Usage Guidelines

The address must be unique on the router.

## Examples

The following example sets up an NSAP address for the local video dial peer designated as 10:

```
dial-peer video 10 videocodec
 nsap 47.0091810000000002F26D4901.333333333332.02
```

## Related Commands

| Command | Description |
|---|---|
| **dial -peer video** | Defines a video ATM dial peer for a local or remote video codec, specifies video-related encapsulation, and enters dial-peer configuration mode. |
| **show dial -peer video** | Displays dial-peer configuration. |

# null-called-number

To substitute a user-defined number as the called number IE when an incoming H.323 setup message does not contain a called number IE, use the **null-called-number** command in voice service H.323 configuration mode. To disable the addition of the number used as the called number IE, use the **no** form of this command.

**null-called-number override** *string*
**no null-called-number**

## Syntax Description

| | |
|---|---|
| **override** *string* | Specifies the user-defined series of digits for the E.164 or private dialing plan telephone number when the called number IE is missing from the H.323 setup message. Valid entries are the digits 0 through 9. |

## Command Default

The command behavior is disabled. H.323 setup messages missing the called number IE are disconnected.

## Command Modes

Voice service h323 configuration (conf-serv-h323)

## Command History

| Release | Modification |
|---|---|
| 12.4(22)YB | This command was introduced. |
| 15.0(1)M | This command was integrated into Cisco IOS Release 15.0(1)M. |

## Usage Guidelines

For a call connection to be completed the incoming H.323 setup messages must include the called number IE and the E.164 destination address. Calls lacking called number IE are disconnected. The null-called-number is a user-defined number used when the called number IE is missing to complete the call.

## Examples

The following example shows the number 4567 configured as the user-defined number used to complete a call when the H.323 setup message is missing the called number IE:

```
Router(conf-serv-h323)# null-called-number override 4567
```

# numbering-type

To match on a number type for a dial-peer call leg, use the **numbering-type**command in dial-peer configurationmode. To remove the numbering type for a dial-peer call leg, use the **no** form of this command.

**numbering-type** {**international** | **abbreviated** | **national** | **network** | **reserved** | **subscriber** | **unknown**}
**no numbering-type** {**international** | **abbreviated** | **national** | **network** | **reserved** | **subscriber** | **unknown**}

## Syntax Description

| | |
|---|---|
| **international** | International numbering type. |
| **abbreviated** | Abbreviated numbering type. |
| **national** | National numbering type. |
| **network** | Network numbering type. |
| **reserved** | Reserved numbering type. |
| **subscriber** | Subscriber numbering type. |
| **unknown** | Numbering type unknown. |

## Command Default

No default behaviors or values

## Command Modes

Dial-peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(7)XR1 | This command was introduced on the Cisco AS5300. |
| 12.0(7)XK | This command was implemented as follows:<br>• VoIP: Cisco 2600 series, Cisco 3600 series, Cisco MC3810<br>• VoFR: Cisco 2600 series, Cisco 3600 series, Cisco MC3810<br>• VoATM: Cisco 3600 series, Cisco MC3810 |
| 12.1(1)T | This command was integrated into Cisco IOS Release 12.1(1)T and implemented as follows:<br>• VoIP: Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco 7200 series, Cisco 7500 series |
| 12.1(2)T | This command was implemented as follows:<br>• VoIP: Cisco MC3810<br>• VoFR: Cisco 2600 series, Cisco 3600 series, Cisco MC3810<br>• VoATM: Cisco 3600 series, Cisco MC3810 |

| Release | Modification |
|---|---|
| 12.2(2)XB1 | This command was implemented on the Cisco AS5850. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T. |

## Usage Guidelines

Thiscommand is supported for POTS, VoIP, VoFR, and VoATM dial peers. The numbering type options are implemented as defined by the ITU Q.931 specification.

## Examples

The following example shows how to configure a POTS dial peer for network usage:

```
dial-peer voice 100 pots
 numbering-type network
```

The following example shows how to configure a VoIP dial peer for subscriber usage:

```
dial-peer voice 200 voip
 numbering-type subscriber
```

## Related Commands

| Command | Description |
|---|---|
| **rule** | Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls. |
| **show translation -rule** | Displays the contents of all the rules that have been configured for a specific translation name. |
| **test translation -rule** | Tests the execution of the translation rules on a specific name-tag. |
| **translate** | Applies a translation rule to a calling party number or a called party number for incoming calls. |
| **translate -outgoing** | Applies a translation rule to a calling party number or a called party number for outgoing calls. |
| **translation -rule** | Creates a translation name and enters translation-rule configuration mode. |
| **voip -incoming translation-rule** | Captures calls that originate from H.323-compatible clients. |

# num-exp

To define how to expand a telephone extension number into a particular destination pattern, use the **num-exp**command in global configuration mode. To remove the configured number expansion, use the no form of this command.

**num-exp** *extension-number* *expanded-number*
**no** **num-exp** *extension-number*

## Syntax Description

| | |
|---|---|
| *extension -number* | One or more digits that define an extension number for a particular dial peer. |
| *expanded -number* | One or more digits that define the expanded telephone number or destination pattern for the extension number listed. |

## Command Default

No number expansion is defined.

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 12.0(3)T | This command was implemented on the Cisco AS5300. |
| 12.0(4)XL | This command was implemented on the Cisco AS5800. |
| 12.0(7)T | This command was integrated into Cisco IOS Release 12.0(7)T. |
| 12.0(7)XK | This command was implemented on the Cisco MC3810. |
| 12.1(2)T | This command was modified. It was integrated into Cisco IOS Release 12.1(2)T. |
| Cisco IOS XE Bengaluru 17.6.1a | Introduced support for YANG models. |

## Usage Guidelines

Use this command to define how to expand a particular set of numbers (for example, a telephone extension number) into a particular destination pattern. With this command, you can bind specific extensions and expanded numbers together by explicitly defining each number, or you can define extensions and expanded numbers using variables. You can also use this command to convert seven-digit numbers to numbers containing fewer than seven digits.

You can configure a maximum of 250 number extensions before the router sends an error message stating that the limit has been reached.

Use a period (.) as a variable or wildcard, representing a single number. Use a separate period for each number that you want to represent with a wildcard--for example, if you want to replace four numbers in an extension with wildcards, type in four periods.

Translation of a number in +E.164 format is not supported if you use the CLI command **num-exp** , although the plus symbol (+) is displayed as a configurable option for the command. As a workaround, it is recommended that you use translation rule to support the +E.164 dial pattern that contains the plus (+) symbol. For a sample of the configuration, see Example.

## Examples

The following is a sample configuration for support of +E.164 number on the Voice Gateway:

```
router(config)#show num-exp
Dest Digit Pattern = '1001'     Translation =
                                '+4001'
router(config)#num-exp 1001 ?
  WORD  Substitution Pattern to Translate Dialed Pat
        to E.164
```

The following example expands the extension number 50145 to the number 14085550145:

```
num-exp 50145 14085550145
```

The following example expands all five-digit extensions beginning with 5 such that the 5 is replaced with the digits 1408555 at the beginning of the extension number:

```
num-exp 5.... 1408555....
```

## Related Commands

| Command | Description |
|---|---|
| **dial -peer terminator** | Designates a special character to be used as a terminator for variable length dialed numbers. |
| **forward -digits** | Specifies which digits to forward for voice calls. |
| **prefix** | Specifies a prefix for a dial peer. |