# Cisco IOS Voice Commands: I

This chapter contains commands to configure and maintain Cisco IOS voice applications. The commands are presented in alphabetical order. Some commands required for configuring voice may be found in other Cisco IOS command references. Use the master index of commands or search online to find these commands.

For detailed information on how to configure these applications and features, refer to the *Cisco IOS Voice Configuration Library*.

# icpif

To specify the Calculated Planning Impairment Factor (ICPIF) for calls sent by a dial peer, use the **icpif** command in dial peer configuration mode. To reset to the default, use the **no** form of this command.

**icpif** *number*

**no icpif**

## Syntax Description

| | |
|---|---|
| *number* | Integer, expressed in equipment impairment factor units, that specifies the ICPIF value. Range is 0 to 55. The default is 20. |

## Command Default

20

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 12.0(7)XK | This command was implemented on the Cisco MC3810. |
| 12.1(2)T | This command was integrated into Cisco IOS Release 12.1(2)T. |
| 12.2(8)T | The *number* default value for this command was changed from 30 to 20. |

## Usage Guidelines

This command is applicable only to VoIP dial peers.

Use this command to specify the maximum acceptable impairment factor for the voice calls sent by the selected dial peer.

## Examples

The following example disables the **icpif** command:

```
dial-peer voice 10 voip
 icpif 0
```

# id

To configure the local identification (ID) for a neighboring border element (BE), use the **id** command in Annex G neighbor border element (BE) configuration mode. To remove the local ID, use the **no** form of this command.

**id** *neighbor-id*

**no id** *neighbor-id*

## Syntax Description

| | |
|---|---|
| *neighbor-id* | ID for a neighboring BE. The identification ID must be an International Alphabet 5 (IA5) string and cannot include spaces. This identifier is local and is not related to the border element ID. |

## Defaults

No default behavior or values

## Command Modes

Annex G neighbor BE 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. This command is not supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 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. |

## Examples

The following example configures the local ID for a neighboring BE. The identifier is 2333.

```
Router(config-annexg-neigh)# id 2333
```

The following example shows the the error response when an undefined neighbor ID is entered:

```
Router(config-annexg-neigh)#no id def
% Entry not valid, id not configured.

To deconfigure id under different neighbor you have to expilicitly go into that neighbor
and deconfigure the id.
```

## Related Commands

| Command | Description |
|---|---|
| **advertise (annex G)** | Controls the type of descriptors that the BE advertises to its neighbors. |
| **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 queries the neighboring BE. |

# idle-voltage

To specify the idle voltage on an Foreign Exchange Station (FXS) voice port, use the **idle-voltage** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

**idle-voltage** {**high** | **low**}

**no idle-voltage**

## Syntax Description

| | |
|---|---|
| **high** | The talk-battery (tip-to-ring) voltage is high (–48V) when the FXS port is idle. |
| **low** | The talk-battery (tip-to-ring) voltage is low (–24V) when the FXS port is idle. |

## Command Default

The idle voltage is –24V

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(4)T | This command was introduced on the Cisco MC3810. |

## Usage Guidelines

Some fax equipment and answering machines require a –48V idle voltage to be able to detect an off-hook condition in a parallel phone.

If the idle voltage setting is **high**, the talk battery reverts to –24V whenever the voice port is active (off hook).

## Examples

The following example sets the idle voltage to –48V on voice port 1/1:

```
voice-port 1/1
 idle-voltage high
```

The following example restores the default idle voltage (–24V) on voice port 1/1:

```
voice-port 1/1
 no idle-voltage
```

## Related Commands

| Command | Description |
|---|---|
| **show voice port** | Displays voice port configuration information. |

# ignore

To configure the North American E&M or E&M MELCAS voice port to ignore specific receive bits, use the **ignore** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

**ignore** {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}

**no ignore** {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}

## Syntax Description

| | |
|---|---|
| **rx-a-bit** | Ignores the receive A bit. |
| **rx-b-bit** | Ignores the receive B bit. |
| **rx-c-bit** | Ignores the receive C bit. |
| **rx-d-bit** | Ignores the receive D bit. |

## Command Default

The default is mode-dependent:

- North American E&M:
  - The receive B, C, and D bits are ignored
  - The receive A bit is not ignored
- E&M MELCAS:
  - The receive A bit is ignored
  - The receive B, C, and D bits are not ignored

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)MA | This command was introduced on the Cisco MC3810. |
| 12.0(7)XK | This command was implemented on the Cisco 2600 series and Cisco 3600 series. |
| 12.1(2)T | This command was integrated into Cisco IOS Release 12.1(2)T. |

## Usage Guidelines

The **ignore** command applies to E&M digital voice ports associated with T1/E1 controllers. Repeat the command for each receive bit to be configured. Use this command with the **define** command.

## Examples

To configure voice port 1/1 to ignore receive bits A, B, and C and to monitor receive bit D, enter the following commands:

```
voice-port 1/1
 ignore rx-a-bit
 ignore rx-b-bit
 ignore rx-c-bit
 no ignore rx-d-bit
```

To configure voice port 1/0/0 to ignore receive bits A, C, and D and to monitor receive bit B, enter the following commands:

```
voice-port 1/0/0
 ignore rx-a-bit
 ignore rx-c-bit
 ignore rx-d-bit
 no ignore rx-b-bit
```

## Related Commands

| Command | Description |
|---|---|
| **condition** | Manipulates the signaling bit pattern for all voice signaling types. |
| **define** | Defines the transmit and receive bits for North American E&M and E&M MELCAS voice signaling. |
| **show voice port** | Displays configuration information for voice ports. |

# ignore (interface)

To configure the serial interface to ignore the specified serial signals as the line up/down indicator, use the **ignore** command in interface configuration mode. To restore the default, use the **no** form of this command.

### DCE Asynchronous Mode

**ignore** [**dtr** | **rts**]

**no ignore** [**dtr** | **rts**]

### DCE Synchronous Mode

**ignore** [**dtr** | **local-loopback** | **rts**]

**no ignore** [**dtr** | **local-loopback** | **rts**]

### DTE Asynchronous Mode

**ignore** [**cts** | **dsr**]

**no ignore** [**cts** | **dsr**]

### DTE Synchronous Mode

**ignore** [**cts** | **dcd** | **dsr**]

**no ignore** [**cts** | **dcd** | **dsr**]

## Syntax Description

| | |
|---|---|
| **dtr** | Specifies that the DCE ignores the Data Terminal Ready (DTR) signal. |
| **rts** | Specifies that the DCE ignores the Request To Send (RTS) signal. |
| **local-loopback** | Specifies that the DCE ignores the local loopback signal. |
| **cts** | Specifies that the DTE ignores the Clear To Send (CTS) signal. |
| **dsr** | Specifies that the DTE ignores the Data Set Ready (DSR) signal. |
| **dcd** | Specifies that the DTE ignores the Data Carrier Detect (DCD) signal. |

## Command Default

The **no** form of this command is the default. The serial interface monitors the serial signal as the line up/down indicator.

## Command Modes

Interface configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(15)ZJ | This command was introduced on the following platforms: Cisco 2610XM, Cisco 2611XM, Cisco 2620XM, Cisco 2621XM, Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3631, Cisco 3660, Cisco 3725, and Cisco 3745 routers. |
| 12.3(2)T | This command was integrated into Cisco IOS Release 12.3(2)T. |

## Usage Guidelines

### Serial Interfaces in DTE Mode

When the serial interface is operating in DTE mode, it monitors the DCD signal as the line up/down indicator. By default, the attached DCE device sends the DCD signal. When the DTE interface detects the DCD signal, it changes the state of the interface to up.

### SDLC Multidrop Environments

In some configurations, such as a Synchronous Data Link Control (SDLC) multidrop environment, the DCE device sends the DSR signal instead of the DCD signal, which prevents the interface from coming up. Use this command to tell the interface to monitor the DSR signal instead of the DCD signal as the line up/down indicator.

## Examples

The following example shows how to configure serial interface 0 to ignore the DCD signal as the line up/down indicator:

```
Router(config)# interface serial 0
Router(config-if)# ignore dcd
```

## Related Commands

| Command | Description |
|---|---|
| **debug serial lead-transition** | Activates the leads status transition debug capability for all capable ports. |
| **show interfaces serial** | Displays information about a serial interface. |

# image encoding

To specify an encoding method for fax images associated with a Multimedia Mail over IP (MMoIP) dial peer, use the **image encoding** command in dial peer configuration mode. To reset to the default, use the **no** form of this command.

**image encoding** {**mh** | **mr** | **mmr** | **passthrough**}

**no image encoding** {**mh** | **mr** | **mmr** | **passthrough**}

## Syntax Description

| | |
|---|---|
| **mh** | Modified Huffman image encoding. This is the IETF standard. |
| **mr** | Modified Read image encoding. |
| **mmr** | Modified Modified Read image encoding. |
| **passthrough** | The image is not modified by an encoding method. |

## Command Default

Passthrough encoding

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(4)XJ | This command was introduced. |
| 12.0(4)T | This command was integrated into Cisco IOS Release 12.0(4)T. |
| 12.1(1)T | This command was integrated into Cisco IOS Release 12.1(1)T. |
| 12.1(5)T | This command was integrated into Cisco IOS Release 12.1(5)T. |
| 12.2(4)T | This command was implemented on the Cisco 1750. |
| 12.2(8)T | This command was implemented on the following platforms: Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745. |

## Usage Guidelines

Use this command to specify an encoding method for e-mail fax TIFF images for a specific MMoIP dial peer. This command applies primarily to the on-ramp MMoIP dial peer. Although you can optionally create an off-ramp dial peer and configure a particular image encoding value for that off-ramp call leg, store-and-forward fax ignores the off-ramp MMoIP setting and sends the file using Modified Huffman encoding.

There are four available encoding methods:

- Modified Huffman (MH)—One-dimensional data compression scheme that compresses data in only one direction (horizontal). Modified Huffman compression does not allow the transmission of redundant data. This encoding method produces the largest image file size.
- Modified Read (MR)—Two-dimensional data compression scheme (used by fax devices) that handles the data compression of the vertical line and that concentrates on the space between lines and within given characters.

- Modified Modified Read (MMR)—Data compression scheme used by newer Group 3 fax devices. This encoding method produces the smallest possible image file size and is slightly more efficient than Modified Read.
- Passthrough—No encoding method is applied to the image—meaning that the image is encoded by whatever encoding method is used by the fax device.

The IETF standard for sending fax TIFF images is Modified Huffman encoding with fine or standard resolution. RFC 2301 requires that compliant receivers support TIFF images with MH encoding and fine or standard resolution. If a receiver supports features beyond this minimal requirement, you might want to configure the Cisco AS5300 universal access server to send enhanced-quality documents to that receiver.

The primary reason to use a different encoding scheme from MH is to save network bandwidth. MH ensures interoperability with all Internet fax devices, but it is the least efficient of the encoding schemes for sending fax TIFF images. For most images, MR is more efficient than MH, and MMR is more efficient than MR. If you know that the recipient is capable of receiving more efficient encodings than just MH, store-and-forward fax allows you to send the most efficient encoding that the recipient can process. For end-to-end closed networks, you can choose any encoding scheme because the off-ramp gateway can process MH, MR, and MMR.

Another factor to consider is the viewing software. Many viewing applications (for example, those that come with Windows 95 or Windows NT) are able to display MH, MR, and MMR. Therefore you should decide, on the basis of the viewing application and the available bandwidth, which encoding scheme is right for your network.

This command applies to both on-ramp and off-ramp store-and-forward fax functions.

## Examples

The following example selects Modified Modified Read as the encoding method for fax TIFF images sent by MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 image encoding mmr
```

## Related Commands

| Command | Description |
|---|---|
| **image resolution** | Specifies a particular fax image resolution for a specific MMoIP dial peer. |

# image resolution

To specify a particular fax image resolution for a specific multimedia mail over IP (MMoIP) dial peer, use the **image resolution** command in dial peer configuration mode. To reset to the default, use the **no** form of this command.

**image resolution** {**fine** | **standard** | **superfine** | **passthrough**}

**no image resolution** {**fine** | **standard** | **superfine** | **passthrough**}

## Syntax Description

| | |
|---|---|
| **fine** | Configures the fax TIFF image resolution to be 204-by-196 pixels per inch. |
| **standard** | Configures the fax TIFF image resolution to be 204-by-98 pixels per inch. |
| **superfine** | Configures the fax TIFF image resolution to be 204-by-391 pixels per inch. |
| **passthrough** | Indicates that the resolution of the fax TIFF image is not altered. |

## Command Default

passthrough

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(4)XJ | This command was introduced. |
| 12.0(4)T | This command was integrated into Cisco IOS Release 12.0(4)T. |
| 12.1(1)T | This command was integrated into Cisco IOS Release 12.1(1)T. |
| 12.1(5)T | This command was integrated into Cisco IOS Release 12.1(5)T. |
| 12.2(4)T | This command was implemented on the Cisco 1750 access router. |
| 12.2(8)T | This command was implemented on the following platforms: Cisco 1751, Cisco 2600, Cisco 3600, Cisco 3725, and Cisco 3745. |

## Usage Guidelines

Use this command to specify a resolution (in pixels per inch) for e-mail fax TIFF images sent by the specified MMoIP dial peer. This command applies primarily to the on-ramp MMoIP dial peer. Although you can optionally create an off-ramp dial peer and configure a particular image resolution value for that off-ramp call leg, store-and-forward fax ignores the off-ramp MMoIP setting and sends the file using fine resolution.

This command enables you to increase or decrease the resolution of a fax TIFF image, thereby changing not only the resolution but also the size of the fax TIFF file. The IETF standard for sending fax TIFF images is Modified Huffman encoding with fine or standard resolution. The primary reason to configure a different resolution is to save network bandwidth.

This command applies to both on-ramp and off-ramp store-and-forward fax functions.

## Examples

The following example selects fine resolution (204-by-196 pixels per inch) for e-mail fax TIFF images associated with MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 image encoding mh
 image resolution fine
```

## Related Commands

| Command | Description |
|---|---|
| **image encoding** | Specifies an encoding method for fax images associated with an MMoIP dial peer. |

# impedance

To specify the terminating impedance of a voice-port interface, use the **impedance** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

**impedance** {**600c** | **600r** | **900c** | **900r** | **complex1** | **complex2** | **complex3** | **complex4** | **complex5** | **complex6**}

**no impedance** {**600c** | **600r** | **900c** | **900r** | **complex1** | **complex2** | **complex3** | **complex4** | **complex5** | **complex6**}

## Syntax Description

| | |
|---|---|
| **600c** | 600 ohms + 2.15uF[1]. |
| **600r** | Resistive 600-ohm termination. |
| **900c** | 900 ohms + 2.15uF[1]. |
| **900r** | Resistive 900-ohm termination. |
| **complex1** | 220 ohms + (820 ohms ǁ 115 nF)[1]. |
| **complex2** | 270 ohms + (750 ohms ǁ 150 nF)[1]. |
| **complex3** | 370 ohms + (620 ohms ǁ 310 nF)[1]. |
| **complex4** | 600r, line = 270 ohms + (750 ohms ǁ 150 nF)[1]. |
| **complex5** | 320 + (1050 ohms ǁ 230 nF), line = 12 Kft[1]. |
| **complex6** | 600r, line = 350 + (1000 ohms ǁ 210 nF)[1]. |

1. The plus symbol (+) indicates serial. The double pipe ( ǁ ) indicates parallel.

**Note** This table represents the full set of impedances. Not all modules support the full set of impedance values shown here. To determine which impedance values are available on your modules, enter impedance ? in the command-line interface to see a list of the values you can configure.

## Command Default

600r

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on Cisco 3600 series. |
| 12.3(7)T | This command was integrated into Cisco IOS Release 12.3(7)T and support was added for the **complex3**, **complex4**, **complex5**, and **complex6** keywords on the Cisco 2600XM series, Cisco 2691, Cisco 2800 series, Cisco 3662 (telco models), Cisco 3700 series, and Cisco 3800 series. |

## Usage Guidelines

Use this command to specify the terminating impedance of analog telephony interfaces. The impedance value must match the specifications from the telephony system to which it is connected. Different countries often have different standards for impedance. CO switches in the United States are predominantly 600r. PBXs in the United States are 600r or 900c.

If the impedance is set incorrectly (if there is an impedance mismatch), a significant amount of echo is generated (which could be masked if the **echo-cancel** command has been enabled). In addition, gains might not work correctly if there is an impedance mismatch.

Configuring the impedance on a voice port changes the impedance on both voice ports of a VPM card. This voice port must be shut down and then opened for the new value to take effect.

## Examples

The following example configures an FXO voice port on the Cisco 3600 series router for an impedance of 600 ohms (real):

```
voice-port 1/0/0
impedance 600r
shutdown/no shutdown
```

The following example configures an E&M voice port on a Cisco 2800 for an impedance of complex3:

```
voice-port 1/1
impedance complex3
shutdown/no shutdown
```

## Related Commands

| Command | Description |
|---|---|
| **voice-port** | Enters voice-port configuration mode. |
| **echo-cancel enable** | Enables the cancellation of voice that is sent out the interface and received back on the same interface. |

# inband-alerting

To enable inband alerting, use the **inband-alerting** command in the SIP user agent configuration mode. To disable inband alerting, use the **no** form of this command.

**inband-alerting**

**no inband-alerting**

## Syntax Description

This command has no arguments or keywords.

## Command Default

Enabled

## Command Modes

SIP user agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced. |
| 12.1(3)T | This command was limited to enabling and disabling inband alerting. |
| 12.2(2)XA | This command was implemented on the Cisco AS5350 and Cisco AS5400. |
| 12.2(2)XB1 | This command was introduced on the Cisco AS5850. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T. |

## Usage Guidelines

If inband alerting is enabled, the originating gateway can open an early media path (upon receiving a 180 or 183 message with a SDP body). Inband alerting allows the terminating gateway or switch to feed tones or announcements before a call is connected. If inband alerting is disabled, local alerting is generated on the originating gateway.

To reset this command to the default value, use the **default** command.

## Examples

The following example disables inband alerting:

```
Router(config)# sip-ua
Router(config-sip-ua)# no inband-alerting
```

## Related Commands

| Command | Description |
|---|---|
| **default** | Sets a command to its default. |
| **exit** | Exits the SIP user agent configuration mode. |
| **max-forwards** | Specifies the maximum number of hops for a request. |
| **no** | Negates a command or set its defaults. |
| **retry** | Configures the SIP signaling timers for retry attempts. |

| Command | Description |
|---|---|
| **timers** | Configures the SIP signaling timers. |
| **transport** | Enables SIP UA transport for TCP/UDP. |

# inbound ttl

To set the inbound time-to-live value, use the **inbound ttl** command in Annex G neighbor service configuration mode. To reset to the default, use the **no** form of this command.

**inbound ttl** *ttl-value*

**no inbound ttl**

## Syntax Description

| | |
|---|---|
| *ttl-value* | Inbound time-to-live (TTL) value, in seconds. Range is 0 to 2147483. When set to 0, the service relationship does not expire. The default is 120. |

## Defaults

120 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. Establishing a service relationship between border element A and border element B entitles A 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 that B establishes with A is designated the "inbound" service relationship. Use this command to indicate the duration of the relationship between border elements that participate in a service relationship.

## Examples

The following example sets the inbound time-to-live value to 420 seconds (7 minutes):

```
Router(config-nxg-neigh-svc)# inbound ttl 420
```

## Related Commands

| Command | Description |
|---|---|
| **access-policy** | Requires that a neighbor be explicitly configured. |
| **outbound retry-interval** | Defines the retry period for attempting to establish the outbound relationship between border elements. |
| **retry interval** | Defines the time between delivery attempts. |
| **retry window** | Defines the total time that a border element attempts delivery. |
| **service-relationship** | Establishes a service relationship between two border elements. |
| **shutdown** | Enables or disables the border element. |

# incoming alerting

To instruct an FXO ground-start voice port to modify its means of detecting an incoming call, use the **incoming alerting** command in voice-port configuration mode. To return to the default call detection method, use the **no** form of this command.

**incoming alerting** {**ring-only**}

**no incoming alerting**

## Syntax Description

| | |
|---|---|
| **ring-only** | Count incoming rings to detect incoming calls to the voice port that should be answered by the router. |

## Command Default

The FXO ground-start voice port detects an incoming call either by detecting the ring voltage applied to the line by the PSTN central office (CO) or by detecting that tip-ground is present for greater than about 7 seconds.

## Command Modes

Voice-port configuration

## Command History

| Cisco IOS Release | Modification |
|---|---|
| 12.4(4)XC | This command was introduced. |

## Usage Guidelines

This command is valid only on FXO ports that have been configured with the **signal ground-start** command.

This command is necessary when two Cisco Unified CallManager Express (Cisco Unified CME) routers are used to provide redundant failover for incoming PSTN FXO ground-start lines. The voice ports for these trunk lines are wired in parallel between the two routers. The primary router is set to answer incoming calls after the first ring by default. The secondary router is set to answer incoming calls after 2 or 3 rings using the **ring number** command in voice-port configuration mode. As long as the primary router is operating, then the secondary router will not see enough rings to trigger it to answer the call. When the primary router is not operating, the secondary router has to be able to detect incoming ring signals so that it can answer calls. The default method of incoming call detection is not appropriate for voice ports on a secondary Cisco Unified CME router. The **incoming alerting ring-only** command must be used to modify the incoming call detection logic so that the voice port counts the number of incoming call rings instead of using the default call detection method.

## Examples

The following example sets ring-only as the detection method for incoming calls on voice port 3/0/0, which is an FXO ground-start voice port.

```
Router(config)# voice-port 3/0/0
Router(config-voiceport)# signal ground-start
Router(config-voiceport)# incoming alerting ring-only
```

## Related Commands

| Command | Description |
|---|---|
| **ring number** | Specifies the maximum number of rings to be detected before an incoming call is answered by the router. |
| **signal** | Specifies the type of signaling for a voice port. |

# incoming called-number (call filter match list)

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

**incoming called-number** [**+**]*string*[**T**]

**no incoming called-number** [**+**]*string*[**T**]

## Syntax Description

| | |
|---|---|
| **+** | (Optional) Character that indicates an E.164 standard number. |
| *string* | Series of digits that specify a pattern for the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters:<br>• The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads.<br>• Comma (,), which inserts a pause between digits.<br>• Period (.), which matches any entered digit (this character is used as a wildcard).<br>• Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.<br>• Plus sign (+), which indicates that the preceding digit occurred one or more times.<br>**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.<br>• Circumflex (^), which indicates a match to the beginning of the string.<br>• Dollar sign ($), which matches the null string at the end of the input string.<br>• Backslash symbol (\), which is followed by a single character, and matches that character. Can be used with a single character with no other significance (matching that character).<br>• Question mark (?), which indicates that the preceding digit occurred zero or one time.<br>• Brackets ( [ ] ), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range.<br>• Parentheses ( ( ) ), which indicate a pattern and are the same as the regular expression rule. |
| **T** | (Optional) Control character that indicates that the **destination-pattern** value is a variable-length dial string. Using this control character enables the router to wait until all digits are received before routing the call. |

## 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 incoming called number 5550123:

```
call filter match-list 1 voice
 incoming called-number 5550123
```

## 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 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 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. |

# incoming called-number (dial peer)

To specify a digit string that can be matched by an incoming call to associate the call with a dial peer, use the **incoming called-number** command in dial-peer configuration mode. To reset to the default, use the **no** form of this command.

**incoming called-number** [**+**]*string*[**T**]

**no incoming called-number** [**+**]*string*[**T**]

## Syntax Description

<table>
<tr><td>+</td><td>(Optional) Character that indicates an E.164 standard number.</td></tr>
<tr><td>string</td><td>Series of digits that specify a pattern for the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters:
<ul>
<li>The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads.</li>
<li>Comma (,), which inserts a pause between digits.</li>
<li>Period (.), which matches any entered digit (this character is used as a wildcard).</li>
<li>Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.</li>
<li>Plus sign (+), which indicates that the preceding digit occurred one or more times.
<br>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.</li>
<li>Circumflex (^), which indicates a match to the beginning of the string.</li>
<li>Dollar sign ($), which matches the null string at the end of the input string.</li>
<li>Backslash symbol (\), which is followed by a single character, and matches that character. Can be used with a single character with no other significance (matching that character).</li>
<li>Question mark (?), which indicates that the preceding digit occurred zero or one time.</li>
<li>Brackets ( [ ] ), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range.</li>
<li>Parentheses ( ( ) ), which indicate a pattern and are the same as the regular expression rule.</li>
</ul></td></tr>
<tr><td>T</td><td>(Optional) Control character that indicates that the destination-pattern value is a variable-length dial string. Using this control character enables the router to wait until all digits are received before routing the call.</td></tr>
</table>

## Command Default

No incoming called number is defined

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 11.3NA | This command was implemented on the Cisco AS5800. |
| 12.0(4)XJ | This command was modified for store-and-forward fax. |
| 12.0(4)T | This command was integrated into Cisco IOS Release 12.0(4)T. |
| 12.0(7)XK | This command was implemented on the Cisco MC3810. |
| 12.1(2)T | This command was integrated into Cisco IOS Release 12.1(2)T. |
| 12.1(5)T | This command was integrated into Cisco IOS Release 12.1(5)T. |
| 12.2(4)T | This command was implemented on the Cisco 1750. |
| 12.2(8)T | This command was implemented on the following platforms: Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745. |

## Usage Guidelines

When a Cisco device is handling both modem and voice calls, it needs to be able to identify the service type of the call—meaning whether the incoming call to the server is a modem or a voice call. When the access server handles only modem calls, the service type identification is handled through modem pools. Modem pools associate calls with modem resources based on the dialed number identification service (DNIS). In a mixed environment, in which the server receives both modem and voice calls, you need to identify the service type of a call by using this command.

If you do not use this command, the server attempts to resolve whether an incoming call is a modem or voice call on the basis of the interface over which the call arrives. If the call comes in over an interface associated with a modem pool, the call is assumed to be a modem call; if a call comes in over a voice port associated with a dial peer, the call is assumed to be a voice call.

By default, there is no called number associated with the dial peer, which means that incoming calls are associated with dial peers by matching calling number with answer address, call number with destination pattern, or calling interface with configured interface.

Use this command to define the destination telephone number for a particular dial peer. For the on-ramp POTS dial peer, this telephone number is the DNIS number of the incoming fax call. For the off-ramp MMoIP dial peer, this telephone number is the telephone number of the destination fax machine.

This command applies to both VoIP and POTS dial peers and to on-ramp and off-ramp store-and-forward fax functions.

This command is also used to provide a matching VoIP dial peer on the basis of called number when fax or modem pass-through with named signaling events (NSEs) is defined globally on a terminating gateway.

You can ensure that all calls will match at least one dial peer by using the following commands:

```
Router(config)# dial-peer voice tag voip
Router(config-dial-peer)# incoming called-number.
```

## Examples

The following example configures calls that come into the router with a called number of 555-0163 as being voice calls:

```
dial peer voice 10 pots
 incoming called-number 5550163
```

The following example sets the number (310) 555-0142 as the incoming called number for MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 incoming called-number 3105550142
```

# incoming calling-number (call filter match list)

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

**incoming calling-number** [**+**]*string*[**T**]

**no incoming calling-number** [**+**]*string*[**T**]

## Syntax Description

| | |
|---|---|
| **+** | (Optional) Character that indicates an E.164 standard number. |
| *string* | Series of digits that specify a pattern for the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters:<br>• The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads.<br>• Comma (,), which inserts a pause between digits.<br>• Period (.), which matches any entered digit (this character is used as a wildcard).<br>• Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.<br>• Plus sign (+), which indicates that the preceding digit occurred one or more times.<br>**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.<br>• Circumflex (^), which indicates a match to the beginning of the string.<br>• Dollar sign ($), which matches the null string at the end of the input string.<br>• Backslash symbol (\), which is followed by a single character, and matches that character. Can be used with a single character with no other significance (matching that character).<br>• Question mark (?), which indicates that the preceding digit occurred zero or one time.<br>• Brackets ( [ ] ), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range.<br>• Parentheses ( ( ) ), which indicate a pattern and are the same as the regular expression rule. |
| **T** | (Optional) Control character that indicates that the **destination-pattern** value is a variable-length dial string. Using this control character enables the router to wait until all digits are received before routing the call. |

## 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 incoming calling number 5550125:

```
call filter match-list 1 voice
 incoming calling-number 5550125
```

## 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 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 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. |

# incoming dialpeer

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

**incoming dialpeer** *tag*

**no incoming dialpeer** *tag*

## Syntax Description

| | |
|---|---|
| *tag* | Digits that define 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 incoming dial peer 12:

```
call filter match-list 1 voice
 incoming 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 port** | Configure debug filtering for the incoming port. |
| **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 calling-number** | Configure debug filtering for outgoing calling numbers. |
| **outgoing dialpeer** | Configure debug filtering for the outgoing dial peer. |
| **outgoing port** | Configure debug filtering for the outgoing port. |
| **show call filter match-list** | Display call filter match lists. |

# incoming media local ipv4

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

**incoming media local ipv4** *ip_address*

**no incoming 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 incoming media on the local voice gateway, which has IP address 192.168.10.255:

```
call filter match-list 1 voice
 incoming 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 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 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. |

# incoming media remote ipv4

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

**incoming media remote ipv4** *ip_address*

**no incoming 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 incoming media on the remote IP device, which has IP address 192.168.10.255:

```
call filter match-list 1 voice
 incoming 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 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 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. |

# incoming port

To configure debug filtering for the incoming port, use the **incoming 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

**incoming port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-no*}

**no incoming 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)

**incoming port** {*slot-number*/*subunit-number*/*port*}

**no incoming port** {*slot-number*/*subunit-number*/*port*}

### Cisco AS5300

**incoming port** *controller-number*:**D**

**no incoming port** *controller-number*:**D**

### Cisco AS5400

**incoming port** *card*/*port*:**D**

**no incoming port** *card*/*port*:**D**

### Cisco AS5800

**incoming port** {*shelf*/*slot*/*port*:**D** | *shelf*/*slot*/*parent*:*port*:**D**}

**no incoming port** {*shelf*/*slot*/*port*:**D** | *shelf*/*slot*/*parent*:*port*:**D**}

### Cisco MC3810

**incoming port** *slot*/*port*

**no incoming port** *slot*/*port*

## Syntax Description

### Cisco 2600, Cisco 3600 Series and Cisco 3700 Series

| | |
|---|---|
| *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. |

### Cisco AS5300

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

### Cisco AS5400

| | |
|---|---|
| *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. |

### Cisco AS5800

| | |
|---|---|
| *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.<br>• T1 or E1 controller on the T1 card —Valid entries are 0 to 11.<br>• T1 controller on the T3 card—Valid entries are 1 to 28. |
| **:***port* | Specifies the value for the *parent* argument. The valid entry is 0. |
| **:D** | Indicates the D channel associated with ISDN PRI. |

### Cisco MC3810

| | |
|---|---|
| *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:<br>• T1—ANSI T1.403 (1989), Telcordia TR-54016.<br>• E1— ITU G.703.<br>• Analog Voice—Up to six ports (FXS, FXO, E & M).<br>• Digital Voice— Single T1/E1 with cross-connect drop and insert, CAS and CCS signaling, PRI QSIG.<br>• Ethernet—Single 10BASE-T.<br>• Serial—Two five-in-one synchronous serial (ANSI EIA/TA-530, EIA/TA-232, EIA/TA-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 incoming port 1/1/1 on a Cisco 3660 voice gateway:

```
call filter match-list 1 voice
 incoming 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. |
| **outgoing port** | Configure debug filtering for the outgoing port. |
| **show call filter match-list** | Display call filter match lists. |

# incoming secondary-called-number

To configure debug filtering for incoming called numbers from the second stage of a two-stage scenario, use the **incoming secondary-called-number** command in call filter match list configuration mode. To disable, use the **no** form of this command.

**incoming secondary-called-number** *string*

**no incoming secondary-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:<br>• 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).<br>• Comma (,), which inserts a pause between digits.<br>• 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).<br>• Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.<br>• Plus sign (+), which indicates that the preceding digit occurred one or more times.<br>**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.<br>• Circumflex (^), which indicates a match to the beginning of the string.<br>• Dollar sign ($), which matches the null string at the end of the input string.<br>• 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).<br>• Question mark (?), which indicates that the preceding digit occurred zero or one time.<br>• 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.<br>• 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

Two-stage dialing occurs when the voice gateway presents a dial-tone before accepting digits. When a voice call comes into the Cisco IOS voice gateway, the voice port on the router is seized inbound by a PBX or CO switch. The voice gateway then presents a dial tone to the caller and collects digits until it can identify an outbound dial-peer. Dial-peer matching is done digit-by-digit whether the digits are dialed with irregular intervals by humans or in a regular fashion by telephony equipment sending the precollected digits. The voice gateway attempts to match a dial-peer after each digit is received.

## Examples

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

```
call filter match-list 1 voice
 incoming secondary-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. |
| **outgoing called-number** | Configure debug filtering for outgoing called numbers. |
| **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. |

# incoming signaling local ipv4

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

**incoming signaling local ipv4** *ip_address*

**no incoming 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 incoming signaling on the local voice gateway, which has IP address 192.168.10.255:

```
call filter match-list 1 voice
 incoming 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 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. |
| **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. |

# incoming signaling remote ipv4

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

**incoming signaling remote ipv4** *ip_address*

**no incoming 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 incoming signaling on the remote IP device, which has IP address 192.168.10.255:

```
call filter match-list 1 voice
 incoming 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. |
| **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. |
| **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. |

# incoming uri

To specify the voice class used to match a VoIP dial peer to the uniform resource identifier (URI) of an incoming call, use the **incoming uri** command in dial peer voice configuration mode. To remove the URI voice class from the dial peer, use the **no** form of this command.

### H.323 Session Protocol

**incoming uri** {**called** | **calling**} *tag*

**no incoming uri** {**called** | **calling**}

### Session Initiation Protocol (SIP) Session Protocol

**incoming uri** {**from** | **request** | **to** | **via**} *tag*

**no incoming uri** {**from** | **request** | **to** | **via**}

## Syntax Description

| | |
|---|---|
| **called** | Destination URI in the H.225 message of an H.323 call. |
| **calling** | Source URI in the H.225 message of an H.323 call. |
| *tag* | Alphanumeric label that uniquely identifies the voice class. This *tag* argument must be configured with the **voice class uri** command. |
| **from** | From header in an incoming SIP Invite message. |
| **request** | Request-URI in an incoming SIP Invite message. |
| **to** | To header in an incoming SIP Invite message. |
| **via** | Via header in an incoming SIP Invite message. |

## Command Default

No voice class is specified.

## Command Modes

Dial peer voice configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 12.3(4)T | This command was introduced. |
| 15.1(2)T | This command was modified. The **via** keyword was included. |

## Usage Guidelines

- Before you use this command, configure the voice class by using the **voice class uri** command.
- The keywords depend on whether the dial peer is configured for SIP with the **session protocol sipv2** command. The **from**, **request**, **to**, and **via** keywords are available only for SIP dial peers. The **called** and **calling** keywords are available only for dial peers using H.323.

- This command applies rules for dial peer matching. Table 29 and Table 30 show the rules and the order in which they are applied when the **incoming uri** command is used. The gateway compares the dial-peer command to the call parameter in its search to match an inbound call to a dial peer. All dial peers are searched based on the first match criterion. Only if no match is found does the gateway move on to the next criterion.

*Table 29 Dial-Peer Matching Rules for Inbound URI in SIP Calls*

| Match Order | Cisco IOS Command | Incoming Call Parameter |
|---|---|---|
| 1 | **incoming uri via** | Via URI |
| 2 | **incoming uri request** | Request-URI |
| 3 | **incoming uri to** | To URI |
| 4 | **incoming uri from** | From URI |
| 5 | **incoming called-number** | Called number |
| 6 | **answer-address** | Calling number |
| 7 | **destination-pattern** | Calling number |
| 8 | **carrier-id source** | Carrier-ID associated with the call |

*Table 30 Dial-Peer Matching Rules for Inbound URI in H.323 Calls*

| Match Order | Cisco IOS Command | Incoming Call Parameter |
|---|---|---|
| 1 | **incoming uri called** | Destination URI in H.225 message |
| 2 | **incoming uri calling** | Source URI in H.225 message |
| 3 | **incoming called-number** | Called number |
| 4 | **answer-address** | Calling number |
| 5 | **destination-pattern** | Calling number |
| 6 | **carrier-id source** | Source carrier-ID associated with the call |

**Note** Calls using an E.164 number, rather than a URI, use the dial-peer matching rules that existed prior to Cisco IOS Release 15.1(2)T. For information, see the*Dial Peer Configuration on Voice Gateway Routers* document, Cisco IOS Voice Configuration Library.

- You can use this command multiple times in the same dial peer with different keywords. For example, you can use **incoming uri called** and **incoming uri calling** in the same dial peer. The gateway then selects the dial peer based on the matching rules described in Table 29 and Table 30.

## Examples

The following example matches on the destination telephone URI in incoming H.323 calls by using the ab100 voice class:

```
dial-peer voice 100 voip
 incoming uri called ab100
```

The following example matches on the incoming via URI for SIP calls by using the ab100 voice class:

```
dial-peer voice 100 voip
 session protocol sipv2
 incoming uri via ab100
```

## Related Commands

| Command | Description |
|---|---|
| **answer-address** | Specifies the calling number to match for a dial peer. |
| **debug voice uri** | Displays debugging messages related to URI voice classes. |
| **destination-pattern** | Specifies the telephone number to match for a dial peer. |
| **dial-peer voice** | Enters dial peer voice configuration mode to create or modify a dial peer. |
| **incoming called-number** | Specifies the incoming called number matched to a dial peer. |
| **session protocol** | Specifies the session protocol in the dial peer for calls between the local and remote router. |
| **show dialplan incall uri** | Displays which dial peer is matched for a specific URI in an incoming voice call. |
| **voice class uri** | Creates or modifies a voice class for matching dial peers to calls containing a SIP or TEL URI. |

# index (voice class)

To define one or more numbers for a voice class called number, or a range of numbers for a voice class called number pool, use the **index** command in voice class configuration mode. To remove the number or range of numbers, use the **no** form of this command.

**index** *number called-number*

**no index** *number called-number*

## Syntax Description

| | |
|---|---|
| *number* | Digits that identify this index. Range is 1 to 2147483647. |
| *called-number* | Specifies a called number, or a range of called numbers, in E.164 format. |

## Command Default

No index is configured.

## Command Modes

Voice class configuration

## Command History

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

## Usage Guidelines

Use this command to define one or more numbers for a voice class called number, or a range of numbers for a voice class called number pool. You can define multiple indexes for any inbound or outbound voice class called number or voice class called number pool.

When defining a range of numbers for a called number pool:

- The range of numbers must be in E.164 format.
- The beginning number and ending number must be the same length.
- The last digit of each number must be 0 to 9.
- Leading '+' (if used) must be defined from in the range of called numbers.

## Examples

The following example shows the configuration for indexes in voice class called number pool 100:

```
voice class called number pool 100
 index 1 4085550100 - 4085550111 (Range of called numbers are 4085550100 up to 4085550111)
 index 2 +3227045000
```

The following example shows configuration for indexes in voice class called number outbound 222:

```
voice class called number outbound 222
 index 1 4085550101
 index 2 4085550102
 index 2 4085550103
```

## Related Commands

| Command | Description |
|---|---|
| **voice class called number** | One or more called numbers configured for a voice class. |

# info-digits

To automatically prepend two information digits to the beginning of a dialed number associated with the given POTS dial peer, use the **info-digits** command in dial-peer configuration mode. To prepend the info-digits with "00" use the **default info-digits** form of this command. To keep the router from automatically prepending the two-digit information numbers to the beginning of the POTS dial peer, use the **no** form of this command.

**info-digits** *xx*

**default info-digits**

**no info-digits**

## Syntax Description

| | |
|---|---|
| *xx* | Specifies the two-digit prefix that the router will automatically prepend to the dialed number for the given POTS dial peer to identify the origin of the call. This value cannot contain any more or less than two digits. Valid values include:<br>• 00—Regular line<br>• 01—4- and 8-party<br>• 06—Hotel or Motel<br>• 07—Coinless<br>• 10—Test call<br>• 27—Coin<br>• 95—Test call<br>**Note** Values 12 through 19 cannot be assigned because of conflicts with international 20 Automatic Identification of Outward listed directory number sent. |

## Defaults

The dialed number is prepended with "00", indicating that the dialed number is a regular line.

## Command Modes

Dial-peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(1)T | This command was introduced. |
| 12.3(7)T | This command was modified. The default behavior was changed to prepend the dialed number the with "00". |

## Usage Guidelines

This command is designed to prepend a pair of information digits to the beginning of the dialed number string for the POTS dial peer that will enable you to dynamically redirect the outgoing call. The **info-digits** command is only available for POTS dial peers tied to a voice-port that corresponds to Feature Group-D (FGD) Exchange Access North American (EANA) signaling that provides specific call services such as emergency 911 calls in the United States. Configuring the info-digit command for other voice-port types is not advised and may yield undesirable results.

## Examples

The following example prepends the information number string 91 to the beginning of the dialed number for POTS dial peer 10:

```
dial-peer voice 10 pots
 info-digits 91
```

# information-type

To select a specific information type for a Voice over IP (VoIP) or plain old telephone service (POTS) dial peer, use the **information-type** command in dial peer configuration mode. To remove the current information type setting, use the **no** form of this command. To return to the default configuration, use the **no** form of this command.

**information-type** {**fax** | **voice** | **video**}

**no information-type**

## Syntax Description

| | |
|---|---|
| **fax** | The information type is set to store-and-forward fax. |
| **voice** | The information type is set to voice. This is the default. |
| **video** | The information type is set to video. |

## Command Default

Voice

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |
| 12.0(4)XJ | This command was modified for store-and-forward fax. |
| 12.0(4)T | This command was integrated into Cisco IOS Release 12.0(4)T. |
| 12.1(1)T | This command was integrated into Cisco IOS Release 12.1(1)T. |
| 12.1(5)T | This command was integrated into Cisco IOS Release 12.1(5)T. |
| 12.2(4)T | This command was implemented on the Cisco 1750. |
| 12.2(8)T | This command was implemented on the following platforms: Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745. |
| 12.4(11)T | The **video** keyword was added. |

## Usage Guidelines

The **fax** keyword applies to both on-ramp and off-ramp store-and-forward fax functions.

## Examples

The following example shows the configuration for information type fax for VoIP dial peer 10:

```
dial-peer voice 10 voip
 information-type fax
```

The following example shows the configuration for information type video for POTS dial peer 22:

```
dial-peer voice 22 pots
 information-type video
```

## Related Commands

| Command | Description |
|---|---|
| **isdn integrate calltype all** | Enables integrated mode (for data, voice, and video) on ISDN BRI or PRI interfaces. |

# inject guard-tone

To play out a guard tone with the voice packet, use the **inject guard-tone** command in voice-class configuration mode. To remove the guard tone, use the **no** form of this command.

**inject guard-tone** *frequency amplitude* [**idle**]

**no inject guard-tone** *frequency amplitude* [**idle**]

## Syntax Description

| | |
|---|---|
| *frequency* | Frequency, in Hz, of the tone to be injected. Range is integers from 1 to 4000. |
| *amplitude* | Amplitude, in dBm, of the tone to be injected. Range is integers from –50 to –3. |
| **idle** | (Optional) Play out the inverse of the guard tone when there are no voice packets. Idle tone and guard tone are mutually exclusive. |

## Command Default

No guard tone is injected.

## Command Modes

Voice-class configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(4)XD | This command was introduced. |
| 12.3(7)T | This command was integrated into Cisco IOS Release 12.3(7)T. |

## Usage Guidelines

The **inject guard-tone** command has an effect on an ear and mouth (E&M) analog or digital voice port only if the signal type for that port is Land Mobile Radio (LMR). The guard tone is played out with the voice packet to keep the radio channel up. Guard tones of 1950 Hz and 2175 Hz can be filtered out before the voice packet is sent from the digital signal processor (DSP) to the network using the **digital-filter** command.

## Examples

The following example configures a guard tone of 1950 Hz and –10 dBm to be played out with voice packets:

```
voice class tone-signal tone1
 inject guard-tone 2175 -30
```

## Related Commands

| Command | Description |
|---|---|
| **digital-filter** | Specifies the digital filter to be used before the voice packet is sent from the DSP to the network. |

# inject pause

To specify a pause between injected tones, use the **inject pause** command in voice-class configuration mode. To remove the pause, use the **no** form of this command.

**inject pause** *index milliseconds*

**no inject pause** *index milliseconds*

## Syntax Description

| | |
|---|---|
| *index* | Order of pauses and tones. Range is integers from 1 to 10. |
| *milliseconds* | Duration, in milliseconds, of the pause between injected tones. Range is integers from 10 to 500. |

## Command Default

*milliseconds*: 0 milliseconds

## Command Modes

Voice-class configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(4)XD | This command was introduced. |
| 12.3(7)T | This command was integrated into Cisco IOS Release 12.3(7)T. |

## Usage Guidelines

The **inject pause** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). Use this command to specify the pause between injected tones specified with the **inject tone** command. Use the *index* argument of this command in conjunction with the *index* argument of the inject tone command to specify the order of the pauses and tones.

## Examples

The following example configures a pause of 100 milliseconds after the injected tone:

```
voice class tone-signal 100
 inject tone 1 2000 0 200
 inject pause 2 100
```

## Related Commands

| Command | Description |
|---|---|
| **inject tone** | Specifies a wakeup or frequency selection tone to be played out before the voice packet. |

# inject tone

To specify a wakeup or frequency selection tone to be played out before the voice packet, use the **inject tone** command in voice-class configuration mode. To remove the tone, use the **no** form of this command.

**inject tone** *index frequency amplitude duration*

**no inject tone** *index frequency amplitude duration*

## Syntax Description

| | |
|---|---|
| *index* | Order of pauses and tones. Range is integers from 1 to 10. |
| *frequency* | Frequency, in Hz, of the tone to be injected. Range is integers from 1 to 4000. |
| *amplitude* | Amplitude, in dBm, of the tone to be injected. Range is integers from –30 to 3. |
| *duration* | Duration, in milliseconds, of the tone to be injected. Range is integers from 10 to 500. |

## Command Default

No tone is injected.

## Command Modes

Voice-class configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(4)XD | This command was introduced. |
| 12.3(7)T | This command was integrated into Cisco IOS Release 12.3(7)T. |

## Usage Guidelines

The **inject tone** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). Use this command with the **inject pause** command to configure wakeup and frequency selection tones. Use the *index* argument of this command in conjunction with the *index* argument of the **inject pause** command to specify the order of the pauses and tones.

If you configure injected tones with this command, be sure to use the **timing delay-voice tdm** command to configure a delay before the voice packet is played out. The delay must be equal to the sum of the durations of the injected tones and pauses in the tone-signal voice class.

## Examples

The following example configures a frequency selection tone to be played out before the voice packet:

```
voice class tone-signal 100
 inject tone 1 1950 3 150
 inject tone 2 2000 0 60
 inject pause 3 60
 inject tone 4 2175 3 150
 inject tone 5 1000 0 50
```

## Related Commands

| Command | Description |
|---|---|
| **inject pause** | Specifies a pause between injected tones. |
| **timing delay-voice tdm** | Specifies the delay before a voice packet is played out. |

# input gain

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

**input gain** {*decibels* | **auto-control** [*auto-dbm*]}

**no input gain** {*decibels* | **auto-control** [*auto-dbm*]}

## Syntax Description

| | |
|---|---|
| *decibels* | Gain, in decibels (dB), to be inserted at the receiver side of the interface. Range is integers from –27 to 16. The default is 0. |
| **auto-control** | Enable automatic gain control. |
| *auto-dbm* | (Optional) Target speech level, in decibels per milliwatt (dBm), to be achieved at the receiver side of the interface. Range is integers from –30 to 3. The default is –9. |

## Command Default

*decibels*: 0 decibels
*auto-dbm*: –9 dBm

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced. |
| 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 is typically a minimum attenuation of –6 dB between phones, especially if echo cancellers are present. Connections are implemented to provide 0 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0 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.

Typical Land Mobile Radio (LMR) signaling systems send 0 dB out and expect –10 dB in. Setting output attenuation to 10 dB is typical. Output attenuation should be adjusted to provide the voice level required by the radio to produce correct transmitter modulation.

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 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

The following example inserts a 3-dB gain at the receiver side of the interface in the Cisco 3600 series router:

```
port 1/0/0
 input gain 3
```

## Related Commands

| Command | Description |
|---|---|
| **output attenuation** | Configures a specific output attenuation value or enables automatic gain control for a voice port. |

# interface (RLM server)

To define the IP addresses of the Redundant Link Manager (RLM) server, use the **interface** command in interface configuration mode. To disable this function, use the **no** form of this command.

**interface** *name-tag*

**no interface** *name-tag*

## Syntax Description

| | |
|---|---|
| *name-tag* | Name to identify the server configuration so that multiple entries of server configuration can be entered. |

## Command Default

Disabled

## Command Modes

Interface configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(7) | This command was introduced. |

## Usage Guidelines

Each server can have multiple entries of IP addresses or aliases.

## Examples

The following example configures the access-server interfaces for RLM servers “Loopback1” and “Loopback2”:

```
interface Loopback1
 ip address 10.1.1.1 255.255.255.255
 interface Loopback2
 ip address 10.1.1.2 255.255.255.255
rlm group 1
 server r1-server
 link address 10.1.4.1 source Loopback1 weight 4
 link address 10.1.4.2 source Loopback2 weight 3
```

## Related Commands

| Command | Description |
|---|---|
| **clear interface** | Resets the hardware logic on an interface. |
| **clear rlm group** | Clears all RLM group time stamps to zero. |
| **link (RLM)** | Specifies the link preference. |
| **protocol rlm port** | Reconfigures the port number for the basic RLM connection for the whole rlm-group. |
| **retry keepalive** | Allows consecutive keepalive failures a certain amount of time before the link is declared down. |

| Command | Description |
|---|---|
| **server (RLM)** | Defines the IP addresses of the server. |
| **show rlm group statistics** | Displays the network latency of the RLM group. |
| **show rlm group status** | Displays the status of the RLM group. |
| **show rlm group timer** | Displays the current RLM group timer values. |
| **shutdown (RLM)** | Shuts down all of the links under the RLM group. |
| **timer** | Overwrites the default setting of timeout values. |

# interface Dchannel

To specify an ISDN D-channel interface and enter interface configuration mode, use the **interface Dchannel** command in global configuration mode.

**interface Dchannel** *interface-number*

## Syntax Description

| | |
|---|---|
| *interface-number* | Specifies the ISDN interface number.<br>**Note** The *interface-number* argument depends on which controller the **rlm-group** subkeyword in the **pri-group timeslots** controller configuration command uses. For example, if the Redundant Link Manager (RLM) group is configured using the **controller e1 2/3** command, the D-channel interface command will be **interface Dchannel 2/3**. |

## Command Default

No D-channel interface is specified.

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(8)B | This command was introduced. |
| 12.2(15)T | This command was integrated into Cisco IOS Release 12.2(15)T. |

## Usage Guidelines

This command is used specifically in Voice over IP (VoIP) applications that require release of the ISDN PRI signaling time slot for RLM configurations.

## Examples

The following example configures a D-channel interface for a Signaling System 7 (SS7)-enabled shared T1 link:

```
controller T1 1
 pri-group timeslots 1-3 nfas_d primary nfas_int 0 nfas_group 0 rlm-group 0
 channel group 23 timeslot 24
 end

! D-channel interface is created for configuration of ISDN parameters:
interface Dchannel1
 isdn T309 4000
 end
```

## Related Commands

| Command | Description |
|---|---|
| **pri-group timeslots** | Specifies an ISDN PRI group on a channelized T1 or E1 controller, and releases the ISDN PRI signaling time slot for environments that require that SS7-enabled VoIP applications share all slots in a PRI group. |

# interface event-log dump ftp

To enable the gateway to write the contents of the interface event log buffer to an external file, use the **interface event-log dump ftp** command in application configuration monitor mode. To reset to the default, use the **no** form of this command.

**interface event-log dump ftp** *server*[**:***port]/file* **username** *username* **password** [*encryption-type*] *password*

**no interface event-log dump ftp** *server*[**:***port]/file* **username** *username* **password** [*encryption-type*] *password*

## Syntax Description

| | |
|---|---|
| *server* | Name or IP address of FTP server where the file is located. |
| *port* | (Optional) Specific port number on server. |
| *file* | Name and path of file. |
| *username* | Username required to access file. |
| *encryption-type* | (Optional) The Cisco proprietary algorithm used to encrypt the password. Values are 0 or 7. To disable encryption enter 0; to enable encryption enter 7. If you specify 7, you must enter an encrypted password (a password already encrypted by a Cisco router). |
| *password* | Password required to access file. |

## Command Default

Interface event log buffer is not written to an external file.

## Command Modes

Application configuration monitor

## Command History

| Release | Modification |
|---|---|
| 12.3(14)T | This command was introduced to replace the **call application interface event-log dump ftp** command. |

## Usage Guidelines

This command enables the gateway to automatically write the interface event log buffer to the named file when the buffer becomes full. The default buffer size is 4 KB. To modify the size of the buffer, use the **interface event-log max-buffer-size** command. To manually flush the event log buffer, use the **interface dump event-log** command in privileged EXEC mode.

**Note**

- Enabling the gateway to write event logs to FTP could adversely impact gateway memory resources in some scenarios, for example, when:
- The gateway is consuming high processor resources and FTP does not have enough processor resources to flush the logged buffers to the FTP server.
- The designated FTP server is not powerful enough to perform FTP transfers quickly
- Bandwidth on the link between the gateway and the FTP server is not large enough
- The gateway is receiving a high volume of short-duration calls or calls that are failing

You should enable FTP dumping only when necessary and not enable it in situations where it might adversely impact system performance.

## Examples

The following example specifies that interface event log are written to an external file named int_elogs.log on a server named ftp-server:

```
application
    monitor
    interface event-log dump ftp ftp-server/elogs/int_elogs.log username myname password 0
    mypass
```

The following example specifies that application event logs are written to an external file named int_elogs.log on a server with the IP address of 10.10.10.101:

```
application
    monitor
    interface event-log dump ftp 10.10.10.101/elogs/int_elogs.log username myname password
    0 mypass
```

## Related Commands

| Command | Description |
|---|---|
| **call application interface event-log dump ftp** | Enable the gateway to write the contents of the interface event log buffer to an external file. |
| **interface dump event-log** | Flushes the event log buffer for application interfaces to an external file. |
| **interface event-log** | Enables event logging for external interfaces used by voice applications. |
| **interface event-log max-buffer-size** | Sets the maximum size of the event log buffer for each application interface. |
| **interface max-server-records** | Sets the maximum number of application interface records that are saved. |
| **show call application interface** | Displays event logs and statistics for application interfaces. |

# interface event-log error only

To restrict event logging to error events only for application interfaces, use the **interface event-log error-only** command in application configuration monitor mode. To reset to the default, use the **no** form of this command.

**interface event-log error-only**

**no interface event-log error-only**

## Syntax Description

This command has no arguments or keywords.

## Command Default

All events are logged.

## Command Modes

Application configuration monitor

## Command History

| Release | Modification |
|---|---|
| 12.3(14)T | This command was introduced to replace the **call application interface event-log error only** command. |

## Usage Guidelines

This command limits the severity level of the events that are logged; it does not enable logging. You must use this command with the **interface event-log** command, which enables event logging for all application interfaces.

## Examples

The following example enables event logging for error events only:

```
application
    monitor
    interface event-log error-only
```

## Related Commands

| Command | Description |
|---|---|
| **call application interface event-log error-only** | Restricts event logging to error events only for application interfaces. |
| **interface event-log** | Enables event logging for external interfaces used by voice applications. |
| **interface event-log max-buffer-size** | Sets the maximum size of the event log buffer for each application interface. |
| **interface max-server-records** | Sets the maximum number of application interface records that are saved. |
| **show call application interface** | Displays event logs and statistics for application interfaces. |

# interface event-log max-buffer-size

To set the maximum size of the event log buffer for each application interface, use the **interface event-log max-buffer-size** command in application configuration monitor mode. To reset to the default, use the **no** form of this command.

**interface event-log max-buffer-size** *kbytes*

**no interface event-log max-buffer-size**

## Syntax Description

| | |
|---|---|
| *kbytes* | Maximum buffer size, in kilobytes. Range is 1 to 10. Default is 4. |

## Command Default

4 KB

## Command Modes

Application configuration monitor

## Command History

| Release | Modification |
|---|---|
| 12.3(14)T | This command was introduced to replace the **call application interface event-log max-buffer-size** command. |

## Usage Guidelines

If the event log buffer reaches the limit set by this command, the gateway allocates a second buffer of equal size. The contents of both buffers is displayed when you use the **show call application interface** command. When the first event log buffer becomes full, the gateway automatically appends its contents to an external FTP location if the **interface event-log dump ftp** command is used.

A maximum of two buffers are allocated for an event log. If both buffers are filled, the first buffer is deleted and another buffer is allocated for new events (buffer wraps around). If the **interface event-log dump ftp** command is configured and the second buffer becomes full before the first buffer is dumped, event messages are dropped and are not recorded in the buffer.

## Examples

The following example sets the maximum buffer size to 8 KB:

```
application
   monitor
   interface event-log max-buffer-size 8
```

## Related Commands

| Command | Description |
|---|---|
| **call application interface event-log max-buffer-size** | Sets the maximum size of the event log buffer for each application interface. |
| **interface dump event-log** | Flushes the event log buffer for application interfaces to an external file. |
| **interface event-log dump ftp** | Enables the gateway to write the contents of the interface event log buffer to an external file. |

| Command | Description |
|---|---|
| **interface max-server-records** | Sets the maximum number of application interface records that are saved. |
| **show call application interface** | Displays event logs and statistics for application interfaces. |

# interface max-server-records

To set the maximum number of application interface records that are saved, use the **interface max-server-records** command in application configuration monitor mode. To reset to the default, use the **no** form of this command.

**interface max-server-records** *number*

**no interface max-server-records**

## Syntax Description

| | |
|---|---|
| *number* | Maximum number of records to save. Range is 1 to 100. Default is 10. |

## Command Default

10

## Command Modes

Application configuration monitor

## Command History

| Release | Modification |
|---|---|
| 12.3(14)T | This command was introduced to replace the **call application interface max-server-records** command. |

## Usage Guidelines

Only the specified number of records from the most recently accessed servers are kept.

## Examples

The following example sets the maximum saved records to 50:

```
application
    monitor
    interface max-server-records 50
```

## Related Commands

| Command | Description |
|---|---|
| **call application interface max-server-records** | Sets the maximum number of application interface records that are saved. |
| **interface event-log** | Enables event logging for external interfaces used by voice applications. |
| **interface event-log max-buffer-size** | Sets the maximum size of the event log buffer for each application interface. |
| **show call application interface** | Displays event logs and statistics for application interfaces. |

# interface stats

To enable statistics collection for application interfaces, use the **interface stats** command in application configuration monitor mode. To reset to the default, use the **no** form of this command.

**interface stats**

**no interface stats**

## Syntax Description

This command has no arguments or keywords.

## Command Default

Statistics collection is disabled.

## Command Modes

Application configuration monitor

## Command History

| Release | Modification |
|---|---|
| 12.3(14)T | This command was introduced to replace the **call application interface stats** command. |

## Usage Guidelines

To display the interface statistics enabled by this command, use the **show call application interface** command. To reset the interface counters to zero, use the **clear call application interface** command.

## Examples

The following example enables statistics collection for application interfaces:

```
application
    monitor
    interface stats
```

## Related Commands

| Command | Description |
|---|---|
| **call application interface stats** | Enables statistics collection for application interfaces. |
| **clear call application interface** | Clears application interface statistics or event logs. |
| **interface event-log** | Enables event logging for external interfaces used by voice applications. |
| **show call application interface** | Displays event logs and statistics for application interfaces. |
| **stats** | Enables statistics collection for voice applications. |

# ip circuit

To create carrier IDs on an IP virtual trunk group, and create a maximum capacity for the IP group, use the **ip circuit** command. To remove a trunk group or maximum capacity, use the **no** form of the command.

**ip circuit** {**carrier-id** *carrier-name* [**reserved-calls** *reserved*] | **max-calls** *maximum-calls* | **default** {**only** | **name** *carrier-name*}}

**no ip circuit** {**carrier-id** *carrier-name* | **default** {**only** | **name** *carrier-name*}}

## Syntax Description

| | |
|---|---|
| **carrier-id** | Sets the IP circuit associated with a specific carrier. |
| *carrier-name* | Defines an IP circuit using the specified name as the circuit ID. |
| **reserved-calls** *reserved* | (Optional) Specifies the maximum number of calls for the circuit ID. Default value is 200. |
| **max-calls** *maximum-calls* | Sets the number of maximum aggregate H.323 IP circuit carrier call legs. Default value is 1000. |
| **default only** | Creates a single carrier using the default carrier name. |
| **default name** | Changes the default circuit name. |
| *carrier-name* | Default carrier name. |

## Command Default

If this command is not specified, no IP carriers and no maximum call leg values are defined.

## Command Modes

H.323 configuration.

## Command History

| Release | Modification |
|---|---|
| 12.2(13)T3 | This command was introduced. |

## Usage Guidelines

You can use the **ip circuit** command only when no calls are active. You can define multiple carrier IDs, and the ordering does not matter. IP circuit default only is mutually exclusive with defining carriers with circuit carrier id.

If **ip circuit default only** is specified, the maximum calls value is set to 1000.

## Examples

The following example specifies a default circuit and maximum number of calls:

```
voice service voip
 no allow-connections any to pots
 no allow-connections pots to any
 allow-connections h323 to h323
 h323
  ip circuit max-calls 1000
  ip circuit default only
```

The following example specifies a default carrier and incoming source carrier:

```
voice service voip
 no allow-connections any to pots
 no allow-connections pots to any
 allow-connections h323 to h323
  h323
   ip circuit carrier-id AA reserved-calls 200

   ip circuit max-calls 1000
```

## Related Commands

| Command | Description |
|---|---|
| **show crm** | Displays some of the values set by this command. |
| **voice-source group** | Assigns a name to a set of source IP group characteristics, which are used to identify and translate an incoming VoIP call. |

# ip dhcp-client forcerenew

To enable forcerenew-message handling on the DHCP client when authentication is enabled, use the **ip dhcp-client forcerenew** command in global configuration mode. To disable the forced authentication, use the **no** form of this command.

**ip dhcp-client forcerenew**

**no ip dhcp-client forcerenew**

## Syntax Description

This command has no arguments or keywords.

## Command Default

Forcerenew messages are dropped.

## Command Modes

Global configuration (config)

## 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

DHCP forcerenew handling is not enabled until the CLI is configured.

## Examples

The following example shows how to enable DHCP forcerenew-message handling on the DHCP client:

```
Router(config)# ip dhcp-client forcerenew
```

## Related Commands

| Command | Description |
|---|---|
| **ip dhcp client authentication key-chain** | Specifies the key chain to be used in DHCP authentication requests. |
| **ip dhcp client authentication mode** | Specifies the type of authentication to be used in DHCP messages on the interface. |
| **key chain** | Identifies a group of authentication keys for routing protocols. |

# ip precedence (dial peer)

To set IP precedence (priority) for packets sent by the dial peer, use the **ip precedence** command in dial peer configuration mode. To reset to the default, use the **no** form of this command.

**ip precedence** *number*

**no ip precedence** *number*

## Syntax Description

| | |
|---|---|
| *number* | Integer specifying the IP precedence value. Range is 0 to 7. A value of 0 means that no precedence (priority) has been set. The default is 0. |

## Command Default

The default value for this command is zero (0)

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)NA | This command was introduced on the following platforms: Cisco 2500 series, Cisco 3600 series, and Cisco AS5300. |

## Usage Guidelines

Use this command to configure the value set in the IP precedence field when voice data packets are sent over the IP network. This command should be used if the IP link utilization is high and the quality of service for voice packets needs to have a higher priority than other IP packets. This command should also be used if RSVP is not enabled and the user would like to give voice packets a higher priority than other IP data traffic.

This command applies to VoIP peers.

## Examples

The following example sets the IP precedence to 5:

```
dial-peer voice 10 voip
 ip precedence 5
```

# ip qos defending-priority

To configure the Resource Reservation Protocol (RSVP) defending priority value for determining quality of service (QoS), use the **ip qos defending-priority** command in dial peer configuration mode. To disable RSVP defending priority as a QoS factor, use the **no** form of this command.

**ip qos defending-priority** *defending-pri-value*

**no ip qos defending-priority**

## Syntax Description

| | |
|---|---|
| *defending-pri-value* | The RSVP defending priority value for determining QoS priorities. Valid entries are from 0 to 65535. |

## Command Default

The RSVP defending priority value is disabled and is not a factor in determining QoS.

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 12.4(22)T | This command was introduced. |

## Usage Guidelines

To configure the RSVP defending priority value, use the **ip qos defending-priority** command in dial peer configuration mode. The defending priority value is passed to the QoS module during reservation initiation. In a situation where there is not enough bandwidth available to support all calls, this setting enables an existing call to avoid being preempted by a new call unless the preemption priority of the new call is higher than the defending priority of the existing call.

## Examples

The following example shows how to specify the RSVP defending priority value:

```
dial-peer voice 100 voip
 ip qos defending-priority 1111
```

## Related Commands

| Command | Description |
|---|---|
| **acc-qos** | Defines the acceptable QoS for inbound and outbound calls on a VoIP dial peer. |
| **ip qos dscp** | Configures the DSCP value for QoS. |
| **ip qos policy-locator** | Configures the application ID of RSVP. |
| **ip qos preemption-priority** | Configures the RSVP preemption priority. |
| **ip rsvp policy preempt** | Enables RSVP to take bandwidth from lower-priority reservations and give it to new, higher-priority reservations. |
| **req-qos** | Requests a particular QoS using RSVP to be used in reaching a specified dial peer in VoIP. |
| **show-sip-ua calls** | Displays the active UAC and UAS information for SIP calls on a Cisco IOS device. |
| **voice-class sip rsvp-fail-policy** | Configures RSVP failure policies. |

# ip qos dscp

To configure the differentiated services code point (DSCP) value for quality of service (QoS), use the **ip qos dscp** command in dial peer configuration mode. To disable DSCP as a QoS factor, set the DSCP value to **default** (which sets the value to the 000000 bit pattern). To set DSCP values to their default settings, use the **no** form of this command.

**ip qos dscp** {*dscp-value* | *set-af* | *set-cs* | **default** | **ef**} {**signaling** | **media** [**rsvp-pass** | **rsvp-fail**] | **video** [**rsvp-none** | **rsvp-pass** | **rsvp-fail**]}

**no ip qos dscp** {*dscp-value* | *set-af* | *set-cs* | **default** | **ef**} {**signaling** | **media** [**rsvp-pass** | **rsvp-fail**] | **video** [**rsvp-none** | **rsvp-pass** | **rsvp-fail**]}

## Syntax Description

| | |
|---|---|
| *dscp-value* | DSCP value. Valid entries are from 0 to 63. |
| *set-af* | An assured forwarding bit pattern as the DSCP value:<br>• **af11**—bit pattern 001010<br>• **af12**—bit pattern 001100<br>• **af13**—bit pattern 001110<br>• **af21**—bit pattern 010010<br>• **af22**—bit pattern 010100<br>• **af23**—bit pattern 010110<br>• **af31**—bit pattern 011010<br>• **af32**—bit pattern 011100<br>• **af33**—bit pattern 011110<br>• **af41**—bit pattern 100010<br>• **af42**—bit pattern 100100<br>• **af43**—bit pattern 100110 |
| *set-cs* | Class-selector code point as the DSCP value:<br>• **cs1**—code point 1 (precedence 1)<br>• **cs2**—code point 2 (precedence 2)<br>• **cs3**—code point 3 (precedence 3)<br>• **cs4**—code point 4 (precedence 4)<br>• **cs5**—code point 5 (precedence 5)<br>• **cs6**—code point 6 (precedence 6)<br>• **cs7**—code point 7 (precedence 7) |
| **default** | Specifies the default bit pattern 000000 as the DSCP value. |
| **ef** | Specifies the expedited forwarding bit pattern 101110 as the DSCP value. |
| **signaling** | Specifies that the DSCP value applies to signaling packets. |
| **media** | Specifies that the DSCP value applies to media packets (voice and fax). |
| **rsvp-pass** | (Optional) Specifies that the DSCP value applies to packets with successful Resource Reservation Protocol (RSVP) reservations. |
| **rsvp-fail** | (Optional) Specifies that the DSCP value applies to packets (media or video) with failed RSVP reservations. |
| **video** | Specifies that the DSCP value applies to video packets. This option is valid only for Cisco Unified Communications Manager Express (Cisco Unified CME) on a Cisco Unified Border Element. |
| **rsvp-none** | (Optional) Specifies that the DSCP value applies to video packets with no RSVP reservations (valid only for video packets.) |

## Command Default

The DSCP default values are as follows**:**

- The default DSCP value for all signaling packets is **af31**.
- The default DSCP value for all media (voice and fax) packets is **ef**.
- The default DSCP value for all video packets is **af41**.

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 12.2(2)T | This command was introduced. It replaced the **ip precedence** (dial peer) command |
| 12.3(4)T | Keywords were added to support DSCP configuration for video streams. |
| 12.4(22)T | Keywords were added to apply a DSCP value to media (voice and fax) packets with a specified (successful or failed) RSVP connection. |

## Usage Guidelines

To configure voice, signaling, and video traffic priorities, use the **ip qos dscp** command in dial peer configuration mode. The recommended value for media (voice and fax) packets is **ef;** for signaling packets, the recommended value is **af31**; and for video packets, it is **af41** (all defaults).

Additionally, before you can specify RSVP QoS, you must first use the **ip rsvp bandwidth** command to enable RSVP on the IP interface.

## Examples

The following example shows how to set the DSCP value to a class-selector code point value of 1 and apply that DSCP setting to media (voice and fax) payload packets with no RSVP configured:

```
dial-peer voice 1 voip
 ip qos dscp cs1 media
```

The following example shows how to set the DSCP value to the expedited forwarding bit pattern and apply that DSCP setting to media (voice and fax) payload packets with a successful RSVP connection:

```
dial-peer voice 1 voip
ip qos dscp ef media rsvp-pass
```

The following example shows how to set the DSCP value to an assured forwarding code point value of 22 and apply that DSCP setting to all signaling packets:

```
dial-peer voice 1 voip
ip qos dscp af22 signaling
```

The following example shows how to set the DSCP value to an assured forwarding code point value of 43 and apply that DSCP setting to video packets with a successful RSVP connection:

```
dial-peer voice 100 voip
 ip qos dscp af43 video rsvp-pass
```

## Related Commands

| Command | Description |
|---|---|
| **call rsvp-sync** | Enables synchronization between RSVP signaling and the voice signaling protocol. |
| **ip qos defending-priority** | Configures the RSVP defending priority value. |
| **ip qos policy-locator** | Configures the application ID of RSVP. |
| **ip qos preemption-priority** | Configures the RSVP preemption priority value. |

| Command | Description |
|---|---|
| **ip rsvp bandwidth** | Enables RSVP for IP on an interface. |
| **ip rsvp signalling dscp** | Configures the DSCP settings to be used on RSVP messages on an interface. |

# ip qos policy-locator

To configure a quality of service (QoS) policy-locator (application ID) used to deploy Resource Reservation Protocol (RSVP) policies for specifying bandwidth reservations on Cisco IOS Session Initiation Protocol (SIP) devices, use the **ip qos policy-locator** command in dial peer configuration mode. To delete an application policy, use the **no** form of this command.

**ip qos policy-locator** {**video** | **voice**} [**app** *app-string*] [**guid** *guid-string*] [**sapp** *subapp-string*] [**ver** *version-string*]

**no ip qos policy-locator** {**video** | **voice**} [**app** *app-string*] [**guid** *guid-string*] [**sapp** *subapp-string*] [**ver** *version-string*]

## Syntax Description

| | |
|---|---|
| **video** | Specifies that the application ID applies to RSVP for video streams. |
| **voice** | Specifies that the application ID applies to RSVP for voice streams. |
| **app** | (Optional) Specifies an application. |
| *app-string* | Application ID. Consists of 1 to 31 alphanumeric characters. |
| **guid** | (Optional) Specifies a globally unique identifier (GUID). |
| *guid-string* | GUID. Consists of 1 to 31 alphanumeric characters. |
| **sapp** | (Optional) Specifies a subapplication. |
| *sapp-string* | Subapplication ID. Consists of 1 to 31 alphanumeric characters. |
| **ver** | (Optional) Specifies a version. |
| *ver-string* | Version ID. Consists of 1 to 15 alphanumeric characters. |

## Command Default

No policy is specified.

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 12.4(22)T | This command was introduced. |

## Usage Guidelines

In Cisco IOS software, the RSVP can process and accept requests by referring to multiple bandwidth pools. To enhance the granularity of local policy match criteria on Cisco IOS SIP devices, bandwidth pools can include policies based on application IDs. You can use these application-specific IDs to reserve bandwidth for each until specified bandwidth limits are reached.

To prevent one application type from consuming all bandwidth, RFC 2872, *Application and Sub Application Identity Policy Element for Use with RSVP*, allows for the creation of separate bandwidth reservation pools. For example, an RSVP reservation pool can be created for voice traffic and another for video traffic so that reservations tagged with these application IDs can then be matched to the interface bandwidth pools using RSVP local policies. To limit bandwidth per application, though, you must configure a bandwidth limit for each application and configure each with a reservation flag that associates the application with the appropriate bandwidth limit.

Before you can configure bandwidth limits for any application-specific policy, however, you must create application IDs. To create application IDs (application-specific reservation profiles), use the **ip qos policy-locator** command in dial peer configuration mode. After creating the necessary application IDs, you can then use the appropriate commands listed in the "Related Commands" section to configure bandwidth reservation. However, this feature is available only on supported devices that are running Cisco IOS Release 12.4(22)T or a later release.

For more information about configuring SIP RSVP features, see the "Configuring SIP RSVP Features" chapter in the *Cisco IOS SIP Configuration Guide*. For more general information about the application-specific policy feature, see the "Configuring RSVP" chapter in the RSVP section of the "Signaling" part in the *Cisco IOS Quality of Service Solutions Configuration Guide*.

## Examples

The following example shows how to configure a policy for the application ID:

```
dial-peer voice 100 voip
 ip qos policy-locator voice app MyApp1 sapp MySubApp4
```

## Related Commands

| Command | Description |
|---|---|
| **acc-qos** | Defines the acceptable QoS for inbound and outbound calls on a VoIP dial peer. |
| **handle-replaces** | Configures fallback to legacy handling of SIP INVITE. |
| **ip qos defending-priority** | Configures the RSVP defending priority value. |
| **ip qos dscp** | Sets the DSCP value for QoS. |
| **ip qos preemption-priority** | Configures the RSVP preemption priority value. |
| **ip rsvp bandwidth** | Enables RSVP for IP on an interface. |
| **ip rsvp policy default-reject** | Configures blocking or passing of all messages that do not match any existing RSVP policies. |
| **ip rsvp policy identity** | Defines RSVP application IDs used to deploy RSVP policies. |
| **ip rsvp policy preempt** | Enables RSVP to take bandwidth from lower-priority reservations and give it to new, higher-priority reservations. |
| **maximum (local policy)** | Configures a local policy that limits RSVP resources. |
| **preempt-priority** | Configures RSVP QoS priorities to be inserted into PATH and RESV messages when they are not signaled from an upstream or downstream neighbor or local client application. |
| **req-qos** | Requests a particular QoS using RSVP to be used in reaching a specified dial peer in VoIP. |
| **show sip-ua calls** | Displays the active UAC and UAS information on SIP calls. |
| **voice-class sip rsvp-fail-policy** | Specifies the action that takes place when RSVP negotiation fails. |

# ip qos preemption-priority

To configure the Resource Reservation Protocol (RSVP) preemption priority value for determining quality of service (QoS), use the **ip qos preemption-priority** command in dial peer configuration mode. To disable RSVP preemption priority as a QoS factor, use the **no** form of this command.

**ip qos preemption-priority** *preemption-pri-value*

**no ip qos preemption-priority**

## Syntax Description

| | |
|---|---|
| *preemption-pri-value* | The RSVP preemption priority value for determining QoS priorities. Valid entries are from 0 to 65535. |

## Command Default

The RSVP preemption priority value is disabled and is not a factor in determining QoS.

## Command Modes

Dial peer configuration (config-dial-peer)

## Command History

| Release | Modification |
|---|---|
| 12.4(22)T | This command was introduced. |

## Usage Guidelines

To configure an RSVP preemption priority value, use the **ip qos preemption-priority** command in dial peer configuration mode. The preemption priority value is passed to the QoS module during reservation initiation. In a situation where there is not enough bandwidth available to support all calls, this setting enables a new call to preempt an existing call unless the defending priority of the existing call is higher than the preemption priority of the new call.

## Examples

The following example shows how to specify the RSVP preemption priority value:

```
dial-peer voice 100 voip
 ip qos preemption-priority 1111
```

## Related Commands

| Command | Description |
|---|---|
| **acc-qos** | Defines the acceptable QoS for inbound and outbound calls on a VoIP dial peer. |
| **ip qos dscp** | Configures the DSCP value for QoS. |
| **ip qos policy-locator** | Configures the application ID of RSVP. |
| **ip qos defending-priority** | Configures the defending priority value of RSVP. |
| **ip rsvp policy preempt** | Enables RSVP to take bandwidth from lower-priority reservations and give it to new, higher-priority reservations. |
| **req-qos** | Requests a particular QoS using RSVP to be used in reaching a specified dial peer in VoIP. |
| **show-sip-ua calls** | Displays the active UAC and UAS information for SIP calls on a Cisco IOS device. |
| **voice-class sip rsvp-fail-policy** | Configures RSVP failure policies. |

# ip rtcp report interval

To configure the average reporting interval between subsequent Real-Time Control Protocol (RTCP) report transmissions, use the **ip rtcp report interval** command in global configuration mode. To reset to the default, use the **no** form of this command.

**ip rtcp report interval** *value*

**no ip rtcp report interval**

## Syntax Description

| | |
|---|---|
| *value* | Average interval for RTCP report transmissions, in ms. Range is 1 to 65535. Default is 5000. |

## Command Default

5000 ms

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release. |
| 12.2(11)T | This command was implemented on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800. |

## Usage Guidelines

This command configures the average interval between successive RTCP report transmissions for a given voice session. For example, if the *value* argument is set to 25,000 milliseconds, an RTCP report is sent every 25 seconds, on average.

For more information about RTCP, see RFC 1889, *RTP: A Transport Protocol for Real-Time Applications.*

## Examples

The following example sets the reporting interval to 5000 ms:

```
Router(config)# ip rtcp report interval 5000
```

## Related Commands

| Command | Description |
|---|---|
| **debug ccsip events** | Displays all SIP SPI event tracing and traces the events posted to SIP SPI from all interfaces. |
| **timer receive-rtcp** | Enables the RTCP timer and configures a multiplication factor for the RTCP timer interval. |

# ip rtcp sub-rtcp

To specify sub-Real-Time Control Protocol (RTCP) message types, use the **ip rtcp sub-rtcp** command in global configuration mode. To disable the configuration, use the **no** form of this command.

**ip rtcp sub-rtcp** *message-type number*

**no ip rtcp sub-rtcp** *message-type*

## Syntax Description

| | |
|---|---|
| *message-type* | Message type. For more information, use the question mark (?) online help function. |
| *number* | Message number. The range is from 209 to 255. The default is 209.<br>For more information about the numbering syntax for your networking device, use the question mark (?) online help function. |

## Command Default

RTP payload type is set to the default value 209.

## Command Modes

Global configuration (config)

## Command History

| Release | Modification |
|---|---|
| 15.0(1)M | This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M. |

## Examples

The following example shows how to specify sub-RTCP message typess:

```
Router# configure terminal
Router(config)# ip rtcp sub-rtcp message-type 210
```

## Related Commands

| Command | Description |
|---|---|
| **ip rtcp report interval** | Configures the average reporting interval between subsequent RTCP report transmissions. |

# ip udp checksum

To calculate the UDP checksum for voice packets sent by the dial peer, use the **ip udp checksum** command in dial peer configuration mode. To disable this feature, use the **no** form of this command.

**ip udp checksum**

**no ip udp checksum**

## Syntax Description

This command has no arguments or keywords.

## Command Default

Disabled

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on the Cisco 3600 series. |

## Usage Guidelines

Use this command to enable UDP checksum calculation for each of the outbound voice packets. This command is disabled by default to speed up the transmission of the voice packets. If you suspect that the connection has a high error rate, you should enable this command to prevent corrupted voice packets forwarded to the digital signal processor (DSP).

This command applies to VoIP peers.

**Note** To maintain performance and scalability of the Cisco 5850 when using images before 12.3(4)T, enable no more than 10% of active calls with UDP checksum.

## Examples

The following example calculates the UDP checksum for voice packets sent by dial peer 10:

```
dial-peer voice 10 voip
 ip udp checksum
```

## Related Commands

| Command | Description |
|---|---|
| **loop-detect** | Enables loop detection for T1 for Voice over ATM, Voice over Frame Relay, and Voice over HDLC. |

# irq global-request

To configure the gatekeeper to send information-request (IRQ) messages with the call-reference value (CRV) set to zero, use the **irq global-request** command in gatekeeper configuration mode. To disable the gatekeeper from sending IRQ messages, use the **no** form of this command.

**irq global-request**

**no irq global-request**

## Syntax Description

This command has no arguments or keywords.

## Command Default

The gatekeeper sends IRQ messages with the CRV set to zero.

## Command Modes

Gatekeeper configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(11)T | This command was introduced on the Cisco 3600 series. |

## Usage Guidelines

Use this command to disable the gatekeeper from sending an IRQ message with the CRV set to zero when the gatekeeper requests the status of all calls after its initialization. Disabling IRQ messages can eliminate unnecessary information request response (IRR) messages if the reconstruction of call structures can be postponed until the next IRR or if the call information is no longer required because calls are terminated before the periodic IRR message is sent. Disabling IRQ messages is advantageous if direct bandwidth control is not used in the gatekeeper.

## Examples

The following example shows that IRQ messages are not sent from the gatekeeper:

```
.
.
.
 lrq reject-resource-low
 no irq global-request
 timer lrq seq delay 10
 timer lrq window 6
 timer irr period 6
 no shutdown
.
.
.
```

## Related Commands

| Command | Description |
|---|---|
| **timer irr period** | Configures the IRR timer. |