icpif through irq global-request

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 0to 55. The default is 20.

Command Default

20

Command Modes

Dial-peer configuration (config-dial-peer)

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.

Command Default

No default behavior or values

Command Modes

Annex G neighbor BE configuration (config-annexg-neigh)

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 a 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 (config-voiceport)

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 (config-voiceport)

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 (config-dial-peer)

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 (config-dial-peer)

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 series, Cisco 3600, Cisco 3725, and Cisco 3745.

Usage Guidelines

Use thiscommand 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[2].
900r	Resistive 900-ohm termination.
complex1	220 ohms + (820 ohms || 115 nF)[3].
complex2	270 ohms + (750 ohms || 150 nF)[4].
complex3	370 ohms + (620 ohms || 310 nF)[5].
complex4	600r, line = 270 ohms + (750 ohms || 150 nF)[6].
complex5	320 + (1050 ohms || 230 nF), line = 12 Kft[7].
complex6	600r, line = 350 + (1000 ohms || 210 nF)[8].

1 The plus symbol (+) indicates serial. The double pipe (||) indicates parallel.
2 The plus symbol (+) indicates serial. The double pipe (||) indicates parallel.
3 The plus symbol (+) indicates serial. The double pipe (||) indicates parallel.
4 The plus symbol (+) indicates serial. The double pipe (||) indicates parallel.
5 The plus symbol (+) indicates serial. The double pipe (||) indicates parallel.
6 The plus symbol (+) indicates serial. The double pipe (||) indicates parallel.
7 The plus symbol (+) indicates serial. The double pipe (||) indicates parallel.
8 The plus symbol (+) indicates serial. The double pipe (||) indicates parallel.

Command Default

600r

Command Modes

Voice-port configuration (config-voiceport)

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 thiscommand 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.

Note The values in the syntax description 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.

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 UA configuration (config-sip-ua)

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.

Command	Description
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.
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.

Command Default

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 thiscommand 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.

Command	Description
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 (config-voiceport)

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: • The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit (this character is used as a wildcard). • Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage. • Plus sign (+), which indicates that the preceding digit occurred one or more times. **Note** The plus sign used as part of a digit string is different from the plus sign that can be used in front of a digit string to indicate that the string is an E.164 standard number. • Circumflex (^), which indicates a match to the beginning of the string. • Dollar sign ($), which matches the null string at the end of the input string. • Backslash symbol (\), which is followed by a single character, and matches that character. Can be used with a single character with no other significance (matching that character). • Question mark (?), which indicates that the preceding digit occurred zero or one time. • Brackets ([ ]), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range. • 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

+	(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: • The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit (this character is used as a wildcard). • Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage. • Plus sign (+), which indicates that the preceding digit occurred one or more times. **Note** The plus sign used as part of a digit string is different from the plus sign that can be used in front of a digit string to indicate that the string is an E.164 standard number. • Circumflex (^), which indicates a match to the beginning of the string. • Dollar sign ($), which matches the null string at the end of the input string. • Backslash symbol (\), which is followed by a single character, and matches that character. Can be used with a single character with no other significance (matching that character). • Question mark (?), which indicates that the preceding digit occurred zero or one time. • Brackets ([ ]), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range. • 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 incoming called number is defined

Command Modes

Dial peer configuration (config-dial-peer)

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.
Cisco IOS XE Release 3.3S	This command was integrated into Cisco IOS XE Release 3.3S.

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: • The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit (this character is used as a wildcard). • Percent sign (%), which indicates that the preceding digit occurred zero or more times; similar to the wildcard usage. • Plus sign (+), which indicates that the preceding digit occurred one or more times. **Note** The plus sign used as part of a digit string is different from the plus sign that can be used in front of a digit string to indicate that the string is an E.164 standard number. • Circumflex (^), which indicates a match to the beginning of the string. • Dollar sign ($), which matches the null string at the end of the input string. • Backslash symbol (\), which is followed by a single character, and matches that character. Can be used with a single character with no other significance (matching that character). • Question mark (?), which indicates that the preceding digit occurred zero or one time. • Brackets ([ ]), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range. • 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.

Command	Description
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.

Command	Description
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.

Command	Description
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*}

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

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.

Syntax Description

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

Syntax Description

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.

Syntax Description

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

Syntax Description

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

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

Command Default

No default behavior or values

Command Modes

Call filter match list configuration

Command History

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

Usage Guidelines

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 5550156:

```
call filter match-list 1 voice
 incoming secondary-called-number 5550156
```

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.

Command	Description
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.

Command	Description
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.

Command	Description
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. The tables below 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 1: 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-numbe r**	Called number
6	**answer-address**	Calling number
7	**destination-pattern**	Calling number
8	**carrier-id source**	Carrier-ID associated with the call

Table 2: 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 the tables above.

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 (config-voice-class)

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 add the two-digit prefix to the beginning of a dialed number string associated with the given POTS dial peer, use the **info-digits** command in dial-peer configuration mode. To specify that the two-digit prefix is "00" use the default info-digits form of this command. To prevent the router from automatically adding the two-digit prefix to the beginning of the POTS dial peer, use the no form of this command.

info-digits *prefix-number*

default info-digits

no info-digits

Syntax Description

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

Command Default

The dialed number string is added with 00, indicating that the dialed number string originates from a regular line.

Command Modes

Dial-peer configuration (config-dialpeer)

Command History

Release	Modification
12.2(1)T	This command was introduced.

Release	Modification
12.3(7)T	This command was modified. The default behavior was changed to add the dialed number string the with 00.

Usage Guidelines

This command adds a two-digit prefix to 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 adds the information number string 91 to the beginning of the dialed number string 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 (config-dial-peer)

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 (config-voice-class)

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 (config-voice-class)

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 (config-voice-class)

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 to enable automatic gain control, use the **input gain** command in voice-port configuration mode. To disable the selected value of the 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	The gain, in decibels (dB), to be inserted at the receiver side of the interface. The range is integers from –6 to 14. The default is 0 decibels.
auto-control	Enables automatic gain control.
auto-dBm	(Optional) The target speech level, in decibels per milliwatt (dBm), to be achieved at the receiver side of the interface. The range is integers from –30 to 3. The default is –9 dBm.

Command Default

Automatic gain control is disabled.

Command Modes

Voice-port configuration (config-voiceport)

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	This command was modified. 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	This command was modified. The **auto-control** keyword and *auto-dBm* argument were added.

Usage Guidelines

A system-wide loss plan must be implemented by using both the **input gain** and **output attenuation** commands. You must consider other equipment (including PBXs) in the system when you create a loss plan. The default value for the **input gain** command assumes that a standard transmission loss plan is in effect; that is, 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 by increasing the output attenuation.

You can increase the gain of a signal coming into the device. If the voice level is too low, use the **input gain** command to increase the input gain.

Typical Land Mobile Radio (LMR) signaling systems send 0 dB out and expect –10 dB in. Setting the 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. Radio network loss and other environmental factors could cause the speech level arriving at a device from an LMR system to 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 shows insertion of 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.

intensity

To configure the intensity or depth of the noise reduction process, use the **intensity** command in media profile configuration mode. To disable the configuration, use the **no** form of this command.

intensity *level*

no intensity *level*

Syntax Description

level	Intensity level. The range is from 0 to 6.

Command Default

Intensity of noise reduction is not configured.

Command Modes

Media profile configuration (cfg-mediaprofile)

Command History

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

Usage Guidelines

Use the **intensity** command to configure the intensity or depth of the noise reduction process. You must create a media profile for noise reduction and then configure the intensity level.

Examples

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

```
Device> enable
Device# configure terminal
Device(config)# media profile nr 200
Device(cfg-mediaprofile)# intensity 2
Device(cfg-mediaprofile)# end
```

Related Commands

Command	Description
media profile nr	Creates a media profile to configure noise reduction parameters.
noisefloor	Configures the noise level, in dBm, above which NR will operate.

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 (config-if)

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.

Command	Description
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.
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. **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 (config)

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.

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 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.
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.

Command	Description
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 address trusted

To set up toll-fraud prevention support on a device, use the **ip address trusted** command in voice-service configuration mode. To disable the setup, use the **no** form of this command.

ip address trusted {**authenticate** | **call-block cause** *code* | **list**}

no ip address trusted {**authenticate** | **call-block cause** | **list**}

Syntax Description

authenticate	Enables IP address authentication on incoming H.323 or Session Initiation Protocol (SIP) trunk calls.
call-block cause *code*	Enables issuing a cause code when an incoming call is rejected on the basis of failed IP address authentication. By default, the device issues a call-reject (21) cause code.
list	Enables manual addition of IPv4 and IPv6 addresses to the trusted IP address list.

Command Default

Toll-fraud prevention support is enabled.

Command Modes

Voice service configuration (conf-voi-serv)

Command History

Release	Modification
15.1(2)T	This command was introduced.

Usage Guidelines

Use the **ip address trusted** command to modify the default behavior of a device, which is to not trust a call setup from a VoIP source. With the introduction of this command, the device checks the source IP address of the call setup before routing the call.

A device rejects a call if the source IP address does not match an entry in the trusted IP address list that is a trusted VoIP source. To create a trusted IP address list, use the **ip address trusted list** command in voice service configuration mode, or use the IP addresses that have been configured using the **session target** command in dial peer configuration mode. You can issue a cause code when an incoming call is rejected on the basis of failed IP address authentication.

Examples

The following example displays how to enable IP address authentication on incoming H.323 or SIP trunk calls for toll-fraud prevention support.:

```
Device(config)# voice service voip
Device(conf-voi-serv)# ip address trusted authenticate
```

The following example displays the number of rejected calls:

```
Device# show call history voice last 1 | inc Disc

DisconnectCause=15
DisconnectText=call rejected (21)
DisconnectTime=343939840 ms
```

The following example displays the error message code and the error description:

```
Device# show call history voice last 1 | inc Error

InternalErrorCode=1.1.228.3.31.0
```

The following example displays the error description:

```
Device# show voice iec description 1.1.228.3.31.0

IEC Version: 1
Entity: 1 (Gateway)
Category: 228 (User is denied access to this service)
Subsystem: 3 (Application Framework Core)
Error: 31 (Toll fraud call rejected)
Diagnostic Code: 0
```

The following example shows how to issue a cause code when an incoming call is rejected on the basis of failed IP address authentication:

```
Device(config)# voice service voip
Device(conf-voi-serv)# ip address trusted call-block cause call-reject
```

The following example displays how to enable the addition of IP addresses to a trusted IP address list:

```
Device(config)# voice service voip
Device(conf-voi-serv)# ip address trusted list
```

Related Commands

Command	Description
debug voip ccapi inout	Traces the execution path through the call control API.
show call history voice	Displays the call history table for voice calls.
show ip address trusted list	Displays a list of valid IP addresses for incoming H.323 or SIP trunk calls.
voice iec syslog	Enables viewing of internal error codes as they are encountered in real time.

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 voice-service configuration (conf-serv-h323)

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 0to 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 (config-dial-peer)

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.

Command	Description
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:	
	• **af11** --bit pattern 001010 • **af12** --bit pattern 001100 • **af13** --bit pattern 001110 • **af21** --bit pattern 010010 • **af22** --bit pattern 010100 • **af23** --bit pattern 010110	• **af31** --bit pattern 011010 • **af32** --bit pattern 011100 • **af33** --bit pattern 011110 • **af41** --bit pattern 100010 • **af42** --bit pattern 100100 • **af43** --bit pattern 100110
set-cs	C lass-selector code point as the DSCP value:	
	• **cs1** --code point 1 (precedence 1) • **cs2** --code point 2 (precedence 2) • **cs3** --code point 3 (precedence 3) • **cs4** --code point 4 (precedence 4)	• **cs5** --code point 5 (precedence 5) • **cs6** --code point 6 (precedence 6) • **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

Release	Modification
12.3(4)T	This command was modified. Keywords were added to support DSCP configuration for video streams.
12.4(22)T	This command was modified. Keywords were added to apply a DSCP value to media (voice and fax) packets with a specified (successful or failed) RSVP connection.
Cisco IOS XE Release 3.3S	This command was integrated into Cisco IOS XE Release 3.3S.

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.

Command	Description
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.

Command	Description
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 (config)

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. 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 (config-dial-peer)

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 AS5850 when using images before Cisco IOS Release 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 (config-gk)

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.