Cisco IOS Voice Commands: I

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

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

icpif

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

icpif *number*

no icpif

Syntax Description

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

Defaults

20

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

This command is applicable only to VoIP dial peers.

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

Examples

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

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

id

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

id *neighbor-id*

no id *neighbor-id*

Syntax Description

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

Defaults

No default behavior or values

Command Modes

Annex G neighbor BE configuration

Command History

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

Examples

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

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

Related Commands

Command	Description
advertise (annex G)	Controls the type of descriptors that the BE advertises to its neighbors.
port	Configures the port number of the neighbor that is used for exchanging Annex G messages.
query-interval	Configures the interval at which the local BE queries the neighboring BE.

idle-voltage

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

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

no idle-voltage

Syntax Description

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

Defaults

The idle voltage is –24V

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

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

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

The **idle-voltage** command applies only to FXS voice ports on Cisco MC3810.

Examples

The following example sets the idle voltage to –48V on voice port 1/1 on a Cisco MC3810:

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

The following example restores the default idle voltage (–24V) on voice port 1/1 on a Cisco MC3810:

```
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.

Defaults

The default is mode-dependent:

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

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

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

Examples

To configure voice port 1/1 on a Cisco MC3810 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 on a Cisco 3600 series router 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.

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.

Defaults

Passthrough encoding

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

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

There are four available encoding methods:

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

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

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

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

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

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

Examples

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

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

Related Commands

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

image resolution

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

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

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

Syntax Description

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

Defaults

passthrough

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

Related Commands

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

impedance

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

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

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

Syntax Description

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

1. The plus symbol (+) indicates serial. The double pipe (||) means in parallel.

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

Defaults

600r

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

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

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

Related Commands

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

inband-alerting

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

inband-alerting

no inband-alerting

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

SIP user agent configuration

Command History

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

Usage Guidelines

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

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

Examples

The following example disables inband alerting:

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

Related Commands

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

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

inbound ttl

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

inbound ttl *ttl-value*

no inbound ttl

Syntax Description

ttl-value	Inbound time-to-live (TTL) value, in seconds. Range is 0 to 4294967295. When set to 0, the service relationship does not expire. The default is 120.

Defaults

120 seconds

Command Modes

Annex G neighbor service configuration

Command History

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

Usage Guidelines

Service relationships are defined to be unidirectional. Establishing a service relationship between border element A and border element B entitles A to send requests to B and expect responses. For B to send requests to A and expect responses, a second service relationship must be established. From A's perspective, the service relationship that B establishes with A is designated the "inbound" service relationship. Use this command to indicate the duration of the relationship between border elements that participate in a service relationship.

Examples

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

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

Related Commands

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

incoming called-number

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*

no incoming called-number *string*

Syntax Description

string	Incoming called telephone number. Valid entries are any series of digits that specify the E.164 telephone number. The default is the calling number pattern.

Defaults

No incoming called number is defined

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

When a Cisco device (such as a Cisco AS5300 universal access server or Cisco AS5800 universal gateway) 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 service 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 555-9262 as being voice calls:

```
dial peer voice 10 pots
 incoming called-number 5559262
```

The following example sets the number (310) 555-9261 as the incoming called number for MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 incoming called-number 3105559261
```

info-digits

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

info-digits *string*

no info-digits

Syntax Description

string	Specifies the two-digit prefix that the router will automatically prepend to the dialed number for the given POTS dial peer. **Note** This string cannot contain any more or any less than two digits.

Defaults

No default behavior or values.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(1)T	This command was introduced on Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series routers and on Cisco AS5300 series universal access servers.

Usage Guidelines

This command is designed to prepend a pair of information digits to the beginning of the dialed number string for the POTS dial peer that will enable you to dynamically redirect the outgoing call. The **info-digits** command is only available for POTS dial peers.

Examples

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

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

information-type

To select a particular information type for a Multimedia Mail over IP (MMoIP) or plain old telephone service (POTS) dial peer, use the **information-type** command in dial-peer configuration mode. To reset to the default, use the **no** form of this command.

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

no information-type {**fax** | **voice**}

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.

Defaults

Voice

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

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

Examples

The following example sets the information type to fax for MMoIP dial peer 10:

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

input gain

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

input gain *decibels*

no input gain *decibels*

Syntax Description

decibels	Gain, in decibels, to be inserted at the receiver side of the interface. Range is integers from –6 to 14. The default is 0.

Defaults

0 decibels

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

A system-wide loss plan must be implemented using both the **input gain** and **output attenuation** commands. Other equipment (including PBXs) in the system must be considered when creating a loss plan. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there must be an attenuation of –6 dB between phones. Connections are implemented to provide –6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0 dB.

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

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

Examples

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

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

The following example inserts a 3-dB gain at the receiver side of the interface in the Cisco MC3810:

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

Related Commands

Command	Description
output attenuation	Configures a specific output attenuation value for a voice port.

intercom (ephone-dn)

To define the directory number for a Cisco IP phone (ephone-dn) that connects with another Cisco IP phone for the intercom feature, use the **intercom** command in ephone-dn configuration mode. To disable this feature, use the **no** form of this command.

intercom *directory-number* [**barge-in** | **no-auto-answer**] [**label** *label*]

no intercom *directory-number*

Syntax Description

directory-number	Telephone number where the intercom calls are placed.
barge-in	(Optional) Allows inbound intercom calls to force an existing call into the call-hold state and allows the intercom call to be answered immediately.
no-auto-answer	(Optional) Disables intercom auto-answer feature.
label	(Optional) Defines a text label for the intercom.
label	(Optional) The actual text label.

Defaults

Intercom functionality is disabled

Command Modes

Ephone-dn configuration

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

This command dedicates a pair of Cisco ephone-dns for use as a "press to talk" two-way intercom between Cisco IP phones. Intercom lines cannot be used in shared line configurations. If an ephone-dn is configured for intercom operation, it must be associated with one Cisco IP phone only. The intercom attribute causes an IP phone line (ephone-dn) to operate in auto-dial fashion for outbound calls and auto-answer-with-mute for inbound calls.

The **barge-in** keyword allows inbound intercom calls to force an existing call into the call-hold state and allows the intercom call to be answered immediately. The **label** keyword defines a text label for the intercom. The **no-auto-answer** keyword creates for the IP phone line a connection that resembles a private line, automatic ringdown (PLAR).

Examples

The following example sets the intercom on Cisco IP phone directory number 1:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number A5001
Router(config-ephone-dn) name "intercom"
Router(config-ephone-dn) intercom A5002 barge-in
```

The following example shows intercom configuration between two Cisco IP phones:

```
ephone-dn 18
 number A5001
 name "intercom"
 intercom A5002 [barge-in]
ephone-dn 19
 number A5002
 name "intercom"
 intercom A5001 [barge-in]
ephone 4
 button 1:2 2:4 3:18
 ephone 5
 button 1:3 2:6 3:19
```

In this example, directory number (ephone-dn) 18 and directory number (ephone-dn) 19 are set as an intercom pair. Directory number (DN) 18 is associated with button 3 of Cisco IP phone (ephone) 4 and directory number (DN) 19 is associated with button number 3 of Cisco IP phone (ephone) 5. Button 3 on both Cisco IP phone 4 and Cisco IP phone 5 are set as a pair to provide intercom service to each other.

The intercom feature acts as a combination speed-dial PLAR and auto-answer with mute. If the barge-in attribute is set on the DN receiving the intercom call, the existing call is forced into the hold state, and the intercom call is accepted. If the phone user has the handset off hook (that is, not in speakerphone mode), the user hears a warning beep, and the intercom call is immediately connected with two-way audio. If the phone user is using speakerphone mode, the intercom connects with the microphone mute activated.

Note Dialing in to an intercom by any caller and auto-dial to a nonintercom destination are not prohibited. Calls to an intercom dn originated by a nonintercom caller triggers auto-answer. To prevent nonintercom originators from manually dialing to an intercom destination, use of the special A, B, C, or D dual-tone multifrequency (DTMF) digits in the intercom phone numbers is recommended because these digits cannot be dialed from a normal phone.

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.

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.

Defaults

Disabled

Command Modes

Interface configuration

Command History

Release	Modification
11.3(7)	This command was introduced.

Usage Guidelines

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

Examples

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

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

Related Commands

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

Command	Description
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**.

Defaults

No D-channel interface is specified.

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

Examples

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

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

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

Related Commands

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

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	Specifies the maximum number of calls for the circuit ID.
reserved	Maximum number of calls. Default value is 200.
max-calls	Sets the number of maximum aggregate H.323 IP circuit carrier call legs.
maximum calls	Maximum number of 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.

Defaults

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

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

Command Modes

H.323 configuration.

Command History

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

Usage Guidelines

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

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 precedence (dial-peer)

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

ip precedence *number*

no ip precedence *number*

Syntax Description

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

Defaults

The default value for this command is zero (0)

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

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

This command applies to VoIP peers.

Examples

The following example sets the IP precedence to 5:

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

ip qos dscp

To set the DSCP for the quality of service, use the **ip qos dscp** command in dial-peer configuration mode. To disable DSCP, use the **no** form of this command.

ip qos dscp [*number* | *set-af* | *set-cs* | **default** | **ef**] [**media** | **signaling**]

no ip qos dscp [*number* | *set-af* | *set-cs* | **default** | **ef**] [**media** | **signaling**]

Syntax Description

number	(Optional) DSCP value. Range is 0 to 63.
set-af	(Optional) Sets DSCP to assured forwarding bit pattern. Acceptable values are as follows: • **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	(Optional) Sets DSCP to class-selector code-point. Acceptable values are as follows: • **cs1**—codepoint 1 (precedence 1) • **cs2**—codepoint 2 (precedence 2) • **cs3**—codepoint 3 (precedence 3) • **cs4**—codepoint 4 (precedence 4) • **cs5**—codepoint 5 (precedence 5) • **cs5**—codepoint 6 (precedence 6) • **cs7**—codepoint 7 (precedence 7)
default	(Optional) Sets DSCP to default bit pattern 000000.
ef	(Optional) Sets DSCP to expedited forwarding bit pattern 101110.
media	(Optional) Applies DSCP to media payload packets.
signaling	(Optional) Applies DSCP to signaling packets.

Defaults

DSCP is set to bit pattern 000000

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(2)T	This command was introduced to replace the **ip precedence** (dial-peer) command.

Usage Guidelines

To configure voice and signaling traffic priorities, use the **ip qos dscp** command.

Recommended values are **ip qos dscp ef media** and **ip qos dscp af31 signaling**.

Examples

The following example sets the DSCP to precedence 1 and applies it to media payload packets.

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

Related Commands

Command	Description
call rsvp-sync	Enables synchronization between RSVP signaling and the voice signaling protocol.
ip rsvp signalling dscp	Specifies the DSCP to be used on all RSVP messages sent on an interface.

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 milliseconds. Range is 1 to 65535. The default is 5000.

Defaults

5000 milliseconds

Command Modes

Global configuration

Command History

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

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, refer to RFC 1889, *RTP: A Transport Protocol for Real-Time Applications.*

Examples

The following example sets the reporting interval to 5000 milliseconds:

```
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 source-address (cm-fallback)

To enable a router to receive messages from Cisco IP phones through the specified IP addresses and ports, use the **ip source-address** command in call-manager-fallback configuration mode. To disable the router from receiving messages from Cisco IP phones, use the **no** form of this command.

ip source-address *ip-address* [**port** *port*] [**any-match** | **strict-match**]

no ip source-address [*ip-address* **port** *port*] [**any-match** | **strict-match**]

Syntax Description

ip-address	Preexisting router IP address, typically one of the addresses of the Ethernet port of the router.
port	(Optional) Port to which the gateway router connects to receive messages from the Cisco IP phones.
port	(Optional) Port number. The default port number is 2000.
any-match	(Optional) Disables strict IP address checking for registration.
strict-match	(Optional) Requires strict IP address checking for registration.

Defaults

Port number: 2000
Server address match: **any-match**

Command Modes

Call-manager-fallback configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

This is a mandatory command, and the fallback subsystem does not start if the IP address is not provided. If the port number is not provided, the default value (2000) is used. The IP address is usually the IP address of the Ethernet port to which the phones are connected.

Use the **any-match** keyword to instruct the router to permit Cisco IP phone registration even when the IP server address used by the phone does not match the IP source address. This option can be used to allow registration of Cisco IP phones on different subnets that have different Dynamic Host Configuration Protocol (DHCP) default router or TFTP server addresses.

Use the **strict-match** keyword to instruct the router to reject Cisco IP phone registration attempts if the IP server address used by the phone does not exactly match the source address. By dividing the Cisco IP phones into groups on different subnets and giving each group different DHCP default-router or TFTP server addresses, this option can be used to restrict the number of Cisco IP phones allowed to register,

This command enables a router to receive messages from Cisco IP phones through the specified IP addresses and port. If the router receives a registration request from a Cisco IP phone, the router in return requests the phone configuration and dial-plan information from the Cisco IP phone. This data is stored locally in the memory of the router and is used to create voice port and dial-plan information. The voice port and dial-plan information is used to handle telephony calls to and from the Cisco IP phone if the Cisco CallManager is unreachable.

Examples

The following example sets the IP source address and port:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# ip source-address 10.0.0.2 port 2002 strict-match
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters call-manager-fallback configuration mode.

ip source-address (telephony-service)

To enable a router to receive messages from Cisco IP phones through specified IP addresses and ports, use the **ip source-address** command in telephony-service configuration mode. To disable the router from receiving messages from Cisco IP phones, use the **no** form of this command.

ip source-address *ip-address* [**port** *port*] [**any-match** | **strict-match**]

no ip source-address [*ip-address* **port** *port*] [**any-match** | **strict-match**]

Syntax Description

ip-address	Preexisting router IP address, typically one of the addresses of the Ethernet port of the router.
port	(Optional) TCP/IP port used for Skinny Protocol.
port	(Optional) Port number. Default is 2000.
any-match	(Optional) Disables strict IP address checking for registration. This is the default.
strict-match	(Optional) Requires strict IP address checking for registration.

Defaults

Port number: 2000
Server address match: **any-match**

Command Modes

Telephony-service configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

This is a mandatory command. The Cisco IOS Telephony Service router does not start if the IP address and the port information are not provided. If the port number is not provided, the default is port 2000. The IP address is usually the IP address of the Ethernet port to which the phones are connected.

Use the **any-match** keyword to instruct the router to permit Cisco IP phone registration, and use the **strict-match** keyword to instruct the router to reject IP phone registration attempts if the IP server address used by the phone does not exactly match the source address.

This command enables a router to receive messages from Cisco IP phones through the specified IP address and port.

This command helps the router to autogenerate the SEPDEFAULT.cnf file, which is stored in the Flash memory of the router. The SEPDEFAULT.cnf file contains the IP address of one of the Ethernet ports of the router to which the phone should register. This file is specific to the router and cannot be shared by multiple routers. You must perform the following step to enable access to the SEPDEFAULT.cnf file:

```
Router# tftp-server flash:SEPDEFAULT.cnf
```

The Flash file system on some routers limits the number of times the Flash file can be written to or modified. After this limit is exceeded, the Flash memory must be manually erased, and the files contained in the Flash file must be reloaded.

This command can write or modify the SEPDEFAULT.cnf file only when parameters are actually changed. The file is not deleted by execution of the **no ip source-address** command. However, the SEPDEFAULT.cnf file can be manually removed using the **delete** command.

If this command is executed with changed parameters after the Flash file write limit is exceeded, the command fails. To see the detailed operation of this command, turn on the **debug ephone detail** command.

Examples

The following example sets the IP source address and port:

```
Router(config)# telephony-service
Router(config-telephony-service)# ip source-address 1.6.21.4 port 2000 strict-match
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.
ephone-dn	Enters ephone-dn configuration mode.
max-dn	Sets the maximum number of directory numbers that can be supported by the router.
max-ephones	Configures the maximum number of Cisco IP phones that can be supported by the router.
telephony-service	Enables Cisco IOS Telephony Service and enters telephony-service configuration mode.
tftp-server	Enables TFTP access to firmware files on the TFTP server.

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.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

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

This command applies to VoIP peers.

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

Examples

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

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

Related Commands

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

irq global-request

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

irq global-request

no irq global-request

Syntax Description

This command has no arguments or keywords.

Defaults

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

Command Modes

Gatekeeper configuration

Command History

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

Usage Guidelines

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

Examples

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

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

Related Commands

Command	Description
timer irr period	Configures the IRR timer.

isdn bind-l3

To configure an ISDN D-channel serial interface for signaling backhaul and associate it with a session set, use the **isdn bind-l3** command in interface configuration mode. To disable signaling backhaul on an ISDN D-channel serial interface, use the **no** form of this command.

isdn bind-l3 *set-name*

no isdn bind-l3

Syntax Description

set-name	Session set with which you are associating a D-channel interface.

Defaults

The ISDN D channel is not configured for signaling backhaul and is not associated with a session set

Command Modes

Interface configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco AS5300.
12.2(4)T	This command was implemented on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was implemented on the Cisco IAD2420 series. 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 following platforms: Cisco AS5350, Cisco AS5400, and Cisco AS5850.

Examples

The following example configures T1 signaling channel serial 0:23 for signaling backhaul and associate the D channel with the session set named “Set1”:

```
Router(config)# interface s0:23
Router(config-if)# isdn bind-L3 set1
Router(config-if)# exit
```

The following example configures E1 signaling channel serial 0:15 for signaling backhaul and associates the D channel with the session set named “Set3”:

```
Router(config)# interface s0:15
Router(config-if)# isdn bind-L3 set3
Router(config-if)# exit
```

isdn bind-l3 ccm-manager

To bind Layer 3 of the ISDN PRI interface of the Media Gateway Control Protocol (MGCP) voice gateway to the Cisco CallManager for PRI Q.931 signaling backhaul support, use the **isdn bind-l3 ccm-manager** command in interface configuration mode. To disable this binding, use the **no** form of this command.

isdn bind-l3 ccm-manager

no isdn bind-l3 ccm-manager

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Interface configuration

Command History

Release	Modification
12.2(2)XN	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco Voice Gateway 200 (Cisco VG200).
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2, and implemented on the Cisco IAD2420.

Usage Guidelines

This command enables ISDN PRI backhaul on an MGCP-enabled voice gateway.

Examples

The following example binds PRI Layer 3 to the Cisco CallManager:

```
isdn bind-l3 ccm-manager
```

isdn bind-l3 iua-backhaul

To specify ISDN backhaul using Stream Control Transmission Protocol (SCTP) for an interface, use the **isdn bind-l3 iua-backhaul** command in interface configuration mode. To disable the backhaul capability, use the **no** form of this command.

isdn bind-l3 iua-backhaul [*application-server-name*]

no isdn bind-l3 iua-backhaul

Syntax Description

application-server-name	(Optional) Name of the application server (AS) to use for backhauling the interface.

Defaults

No default behavior or values

Command Modes

Interface configuration

Command History

Release	Modification
12.2(4)T	This command was introduced.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series. 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 applicable to the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800 in this release.

Examples

The following example shows IUA backhaul on the application server “as1”:

```
interface Serial1/0:23
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 isdn bind-L3 iua-backhaul as1
```

Related Commands

Command	Description
as	Defines an AS for backhaul.
asp	Defines an ASP for backhaul.

isdn contiguous-bchan

To configure contiguous bearer channel handling on an E1 PRI interface, use the **isdn contiguous-bchan** command in interface configuration mode. To disable the contiguous B-channel handling, use the **no** form of this command.

isdn contiguous-bchan

no isdn contiguous-bchan

Syntax Description

This command has no arguments or keywords.

Defaults

Contiguous B channel handling is disabled

Command Modes

Interface configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on the following platforms: Cisco 2500 series, Cisco 3600 series, Cisco 7200, and Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Use this command to specify contiguous bearer channel handling so that B channels 1 through 30, skipping 16, map to time slots 1 through 31. This is available for E1 PRI interfaces only, when the **primary-qsig** switch type option is configured by using the **isdn switch-type** command.

Examples

The following example shows the configuration on the E1 interface of a Cisco 3660 series router E1 interface:

```
interface Serial5/0:15
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn incoming-voice voice
 isdn continuous-bchan
```

Related Commands

Command	Description
isdn switch-type primary-qsig	Configures the **primary-qsig** switch type for PRI support.

isdn gateway-max-interworking

To prevent an H.323 gateway from checking for ISDN protocol compatibility and dropping information elements (IEs) in call messages, use the **isdn gateway-max-interworking** command global configuration mode. To reset to the default, use the **no** form of this command.

isdn gateway-max-interworking

no isdn gateway-max-interworking

Syntax Description

This command has no arguments or keywords.

Defaults

The gateway checks for protocol compatibility.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)XI	This command was introduced.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

If this command is enabled on an originating H.323 gateway, the information elements (IEs) in call messages to the terminating gateway are not checked for end-to-end protocol compatibility. If this command is enabled on a terminating gateway, IEs are not checked in the reverse direction. If this command is not enabled, and the ISDN protocols are not compatible on the originating and terminating gateways, the gateway drops all IEs, including the progress indicator. The gateway then inserts a progress indicator of 1 into all Progress messages.

Examples

The following example enables maximum interworking:

```
isdn gateway-max-interworking
```

isdn global-disconnect

To allow passage of "release" and "release complete" messages over the voice network, use the **isdn global-disconnect** command in interface configuration mode. To disable the passage of these messages, use the **no** form of this command.

isdn global-disconnect

no isdn global-disconnect

Syntax Description

This command has no arguments or keywords.

Defaults

Passage of messages is disabled by default; "release" and "release complete" messages terminate locally by default.

Command Modes

Interface configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, and Cisco MC3810.

Usage Guidelines

Enter this command under the isdn interface with switch type bri-qsig or pri-qsig. Use this command to allow passage of "release" and "release complete" messages end-to-end across the network. This is required for certain types of QSIG PBXs whose software or features require either Facility or User Info IEs in those messages to be passed end-to-end between the PBXs. All QSIG interfaces that connect the PBXs to the routers must have this command enabled. This command is available when using the BRI QSIG or PRI QSIG switch type in either master or slave mode.

Examples

The following example shows the configuration on the T1 PRI interface of a Cisco 3660 series router:

```
interface Serial5/0:23
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-qsig
 isdn global-disconnect
 isdn overlap-receiving
 isdn incoming-voice voice
```

Related Commands

Command	Description
isdn protocol-emulate	Configures the interface to serve as either the QSIG slave or the QSIG master.
isdn switch-type	Configures the switch type for BRI or PRI support.

isdn gtd

To enable generic transparency descriptor (GTD) mapping for information elements (IEs) sent in ISDN Setup messages, use the **isdn gtd** command in interface configuration mode. To disable GTD mapping, use the **no** form of this command.

isdn gtd

no isdn gtd

Syntax Description

This command has no arguments or keywords.

Defaults

GTD mapping is enabled.

Command Modes

Interface configuration

Command History

Release	Modification
12.2(15)T	This command was introduced.

Usage Guidelines

Use the **isdn gtd** command to enable parameter mapping for the following ISDN IEs to corresponding GTD parameters:

- Originating Line Information—OLI
- Bearer Capability—USI and TMR
- Called Party Number—CPN
- Calling Party Number—CGN
- Redirecting Number—RGN, OCN and RNI

The following GTD parameters, which have no corresponding ISDN IEs, are also supported:

- Calling Party Category—CPC
- Forward Call Indicators—FCI
- Protocol Name—PRN

Examples

The following example enables GTD parameter mapping:

```
isdn gtd
```

isdn i-number

To configure several terminal devices to use one subscriber line, use the **isdn i-number** command in interface configuration mode.

isdn i-number *n ldn*

Syntax Description

n	Subscriber line 1, 2, or 3, as specified in the NTT specification.
ldn	LDN assigned to the router POTS port.

Defaults

Each terminal device uses one subscriber line.

Command Modes

Interface configuration

Command History

Release	Modification
12.1.(2)XF	This command was introduced on the Cisco 800 series.

Usage Guidelines

Enter the **interface bri** command before entering this command.

Examples

The following example shows two LDNs configured under BRI interface 0:

```
interface bri0
 isdn i-number 1 5551234
 isdn i-number 2 5556789
 exit
dial-peer voice 1 pots
 destination-pattern 5551234
 exit
dial-peer voice 2 pots
 destination-pattern 5556789
 exit
```

Related Commands

Command	Description
interface bri	Specifies a BRI interface and enters interface configuration mode.

isdn ie oli

To configure the value of the Originating Line Information (OLI) information element (IE) identifier when the gateway receives ISDN signaling from an MCI switch, use the **isdn ie oli** command in interface configuration mode. To disable the OLI IE identifier, use the **no** form of this command.

isdn ie oli *value*

no isdn ie oli *value*

Syntax Description

value	Hexadecimal number specifying the value that indicates OLI information from the MCI switch. Range is 00-7F.

Defaults

This command is disabled.

Command Modes

Interface configuration

Command History

Release	Modification
12.2(15)T	This command was introduced.

Usage Guidelines

Use the **isdn ie oli** command to configure gateway support for the MCI ISDN variant by specifying the IE value that indicates OLI information.

Examples

The following example configures the OLI IE value to a hex value of 7A:

```
isdn ie oli 7A
```

Related Commands

Command	Description
isdn gtd	Enables GTD parameter mapping for ISDN IEs.

isdn network-failure-cause

To specify the cause code to pass to the PBX when a call cannot be placed or completed because of internal network failures, use the **isdn network-failure-cause** command in interface configuration mode. To disable use of this cause code, use the **no** form of this command.

isdn network-failure-cause *value*

no isdn network-failure-cause *value*

Syntax Description

value	Number, from 1 to 127. See Table 23 for a list of failure cause code values.

Defaults

No default behavior or values

Command Modes

Interface configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, and Cisco MC3810.

Usage Guidelines

The PBX can reroute calls based on the cause code returned by the router.

This command allows the original cause code to be changed to the value specified if the original cause code is not one of the following:

- NORMAL_CLEARING (16)
- USER_BUSY (17)
- NO_USER_RESPONDING (18)
- NO_USER_ANSWER (19)
- NUMBER_CHANGED (22)
- INVALID_NUMBER_FORMAT (28)
- UNSPECIFIED_CAUSE (31)
- UNASSIGNED_NUMBER (1)

Table 23 describes the cause codes.

Table 23 ISDN Failure Cause Codes

Failure Cause Code	Meaning
1	Unallocated or unassigned number.
2	No route to specified transit network.
3	No route to destination.

Table 23 ISDN Failure Cause Codes (continued)

Failure Cause Code	Meaning
6	Channel unacceptable.
7	Call awarded and being delivered in an established channel.
16	Normal call clearing.
17	User busy.
18	No user responding.
19	No answer from user (user alerted).
21	Call rejected.
22	Number changed.
26	Nonselected user clearing.
27	Destination out of order.
28	Invalid number format.
29	Facility rejected.
30	Response to status enquiry.
31	Normal, unspecified.
34	No circuit/channel available.
38	Network out of order.
41	Temporary failure.
42	Switch congestion.
43	Access information discarded.
44	Requested channel not available.
45	Preempted.
47	Resources unavailable, unspecified.
49	Quality of service unavailable.
50	Requested facility not subscribed.
52	Outgoing calls barred.
54	Incoming calls barred.
57	Bearer capability not authorized.
58	Bearer capability not available now.
63	Service or option not available, unspecified.
65	Bearer capability not implemented.
66	Channel type not implemented.
69	Requested facility not implemented.
70	Only restricted digital information bearer capability is available.
79	Service or option not implemented, unspecified.
81	Invalid call reference value.
82	Identified channel does not exist.

Table 23 ISDN Failure Cause Codes (continued)

Failure Cause Code	Meaning
83	Suspended call exists, but this call ID does not.
84	Call ID in use.
85	No call suspended.
86	Call with requested call ID is cleared.
88	Incompatible destination.
91	Invalid transit network selection.
95	Invalid message, unspecified.
96	Mandatory information element missing.
97	Message type nonexistent or not implemented.
98	Message not compatible with call state or message type nonexistent or not implemented.
99	Information element nonexistent or not implemented.
100	Invalid information element contents.
101	Message not compatible with call state.
102	Recovery on timer expiry.
111	Protocol error, unspecified.
127	Interworking, unspecified.

Examples

The following example specifies a cause code to pass to a PBX when a call cannot be placed or completed of internal network failures:

```
isdn network-failure-cause 28
```

isdn outgoing display-ie

To enable the display information element to be sent in the outgoing ISDN message if provided by the upper layers, such as voice or modem. To disable the displaying of the information element in the outgoing ISDN message, use the no form of this command.

isdn outgoing display-ie

no isdn outgoing display-ie

Syntax Description

There are no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Interface configuration

Command History

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

Usage Guidelines

The **isdn outoing display-ie** command is direction dependent, such as network-to-user or user-to-network. Not all ISDN switch types support the **isdn outgoing display-ie** command. The following shows the direction dependency by switch type, and this command can be used to override the dependency:

- ETSI (NTT, NET3,and NET5)—Only network-to-user
- DMS—Both ways
- TS014—Only network-to-user
- TS013—Only network-to-user
- 1TR6—Only network-to-user

Note The 4ESS, 5ESS, NI1, and NI2 switch types are not supported in any direction.

Note When the **isdn protocol-emulate** command is switched between network and user, this command reverts to its default value. The **isdn outoing display-ie** command must be enabled again.

Examples

The following is a running configuration, showing how the the **isdn outgoing display-ie** command is used on a specified serial interface:

```
Router# show running-config interface serial0:23
interface Serial0:23
 no ip address
 dialer idle-timeout 999999
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn T310 30000
 isdn outgoing display-ie
```

Related Commands

Command	Description
isdn protocol-emulate	Configures an ISDN data or voice port to emulate network or user functionality.

isdn protocol-emulate (voice)

To emulate the network side of an ISDN configuration for a Net5 switch type, use the **isdn protocol-emulate** interface configuration command in interface configuration mode. To disable ISDN emulation, use the **no** form of this command.

isdn protocol-emulate {**network** | **user**}

no isdn protocol-emulate {**network** | **user**}

Syntax Description

network	Network side of an ISDN configuration.
user	User side of an ISDN configuration.

Defaults

No default behavior or values

Command Modes

Interface configuration mode

Command History

Release	Modification
12.0(3)XG	This command was introduced.
12.1(1)T	This command was introduced.
12.2(4)T	This command was implemented on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)XB	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was implemented on the Cisco IAD2420 series. This command is not supported on the access servers in this release.
12.2(11)T	This command was implemented on the following platforms: Cisco AS5350, Cisco AS5400, and Cisco AS5850.

Usage Guidelines

The current ISDN signaling stack can emulate the ISDN network side, but it does not conform to the specifications of the various switch types in emulating the network side. This command enables the Cisco IOS software to replicate the public switched network interface to a Private Branch Exchange (PBX). This feature is supported only for the PRI Net5 switch type.

Examples

The following example configures the interface (configured for Net5) to emulate the network-side ISDN:

```
Router(config)# int s0:15
Router(config-if)# isdn protocol-emulate network
```

isdn rlm-group

To specify the RLM group number that ISDN will start using, use the**isdn rlm-group** command in interface configuration mode. To disable this function, use the **no** form of this command.

isdn rlm-group *number*

no isdn rlm-group *number*

Syntax Description

number	Number of the RLM group, from 0 to 5.

Defaults

Disabled

Command Modes

Interface configuration

Command History

Release	Modification
12.0(2)T	This command was introduced.

Usage Guidelines

The **isdn rlm-group** command allows Redundant Link Manager (RLM) to be used as a way of transporting the D channel information (signalling) over Ethernet.

Examples

The following example defines RLM group 1:

```
interface Serial0:23
 ip address 10.0.0.1 255.0.0.0
 encapsulation ppp
 dialer map ip 10.0.0.2 name hawaii 1111111
 dialer load-threshold 1 either
 dialer-group 1
 isdn switch-type primary-ni
 isdn incoming-voice modem
 isdn rlm-group 1
 ppp authentication chap
 ppp multilink
 hold-queue 75 in
```

Related Commands

Command	Description
clear interface virtual-access	Resets the hardware logic on an interface.
clear rlm group	Clears all RLM group time stamps to zero.
interface	Defines the IP addresses of the server, configures an interface type, and enters interface configuration mode.
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.

isdn supp-service mcid

To configure an ISDN serial interface for Malicious Caller Identification (MCID), use the **isdn supp-service mcid** command in interface configuration mode. To disable MCID functionality, use the **no** form of this command.

isdn supp-service mcid

no isdn supp-service mcid

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Interface configuration

Command History

Release	Modification
12.2(15)T	This command was introduced.

Usage Guidelines

MCID using the **isdn supp-service mcid** command is valid only at the ISDN interface level. The switch type must be primary-net5 and configured for the user side.

Examples

The following configuration example shows that the primary-net5 switch and timer are configured for MCID:

```
interface serial0:23
 isdn switch-type primary-net5
 ip address 10.10.10.0. 255.255.255.0
 isdn supp-service mcid
 isdn T-activate 5000
```

Related Commands

Command	Description
interface serial	Specifies a serial interface created on a channelized E1 or channelized T1 controller for ISDN PRI, channel-associated signaling, or robbed-bit signaling.
isdn T-activate	Specifies how long the ISDN serial interface must wait for the malicious caller to be identified.

isdn supp-service tbct

To enable ISDN Two B-Channel Transfer (TBCT) on PRI trunks, use the **isdn supp-service tbct** command in interface or trunk group configuration mode. To reset to the default, use the **no** form of this command.

isdn supp-service tbct [**notify-on-clear**]

no isdn supp-service tbct

Syntax Description

notify-on-clear	(Optional) ISDN switch notifies the gateway whenever a transferred call is cleared.

Defaults

TBCT is disabled.

Command Modes

Interface configuration
Trunk-group configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

- This command enables TBCT for a specific PRI when used in interface configuration mode. It configures TBCT for all PRIs in a trunk group when used in trunk-group configuration mode.
- The **notify-on-clear** keyword is necessary for the gateway to track billing. It is supported only for user-side ISDN interfaces. You must configure the ISDN switch to send a notify message when a call is cleared.

Examples

The following example enables TBCT for interface 0:23:

```
interface Serial0:23
 isdn supp-service tbct
```

The following example enables TBCT for trunk group 1:

```
trunk group 1
 isdn supp-service tbct
```

Related Commands

Command	Description
call application voice transfer mode	Specifies the call-transfer behavior of a TCL or VoiceXML application.
show call active voice redirect	Displays information about active calls that are being redirected using RTPvt or TBCT.

Command	Description
tbct clear call	Terminates billing statistics for one or more active TBCT calls.
tbct max call-duration	Sets the maximum duration allowed for a call that is redirected using TBCT.
tbct max calls	Sets the maximum number of active calls that can use TBCT.
trunk group	Enters trunk-group configuration mode to define or modify a trunk group.

isdn T-activate

To specify how long the ISDN serial interface must wait for a malicious caller to be identified, use the **isdn T-activate** command in interface configuration mode. To disable the timer, use the **no** form of this command.

isdn T-active *ms*

no isdn T-active *ms*

Syntax Description

ms	Number of milliseconds that the ISDN serial interface must wait for the malicious caller to be identified. The range is from 1000 to 15000 ms. The default is 4000 ms but 5000 ms is recommended.

Defaults

4000 milliseconds

Command Modes

Interface configuration

Command History

Release	Modification
12.2(15)T	This command was introduced.

Usage Guidelines

This timer is optional when configuring MCID and valid only at the ISDN interface level. The switch type must be primary-net5 and configured for the user side.

Examples

The following example shows the configuration of the timer:

```
interface serial0:23
 isdn switch-type primary-net5
 ip address 10.10.10.0 255.255.255.0
 isdn suppserv mcid
 isdn T-activate 5000
```

Related Commands

Command	Description
interface serial	Specifies a serial interface created on a channelized E1 or channelized T1 controller for ISDN PRI, channel-associated signaling, or robbed-bit signaling.
isdn suppserv mcid	Configures an ISDN serial interface for MCID.

iua

To specify backhaul using Stream Control Transmission Protocol (SCTP) and to enter IDSN User Adaptation Layer (IUA) configuration mode, use the **iua** command in terminal configuration mode.

iua

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.2(4)T	This command was introduced.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series. 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 integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300 and Cisco AS5850.
12.2(15)T	This command was implemented on the Cisco 2420, Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series; and Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 network access server (NAS) platforms.

Usage Guidelines

You must first enter IUA configuration mode to access SCTP configuration mode. First enter IUA configuration mode by using the example below and then enter **sctp** at the Router(config-iua)#prompt to bring up SCTP configuration mode. See the **sctp** command.

Examples

The following example shows how to enter iua configuration mode:

```
Router# configure terminal

Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)# iua
Router(config-iua)#
```

The following example shows how to configure the failover-timer by setting the failover time (in milliseconds) to 1 second for a particular AS:

```
Router(config-iua)# as as5400-3 fail-over-timer 1000
```

The following example configure the number of SCTP streams for this AS to 57, which is the maximum value allowed:

```
Router(config-iua)# as as5400-3 sctp-streams 57
```

Related Commands

Command	Description
isdn bind-L3 iua-backhaul	Specifies ISDN backhaul using SCTP for an interface.
show iua as	Shows information about the current condition of an AS.
show iua asp	Shows information about the current condition of an ASP.

ivr asr-server

To specify the location of an external media server that provides automatic speech recognition (ASR) functionality to voice applications, use the **ivr asr-server** command in global configuration mode. To remove the server location, use the **no** form of this command.

ivr asr-server *url*

no ivr asr-server

Syntax Description

url	Location of the ASR resource on the media server, in URL format.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the following platforms: Cisco 3640, Cisco 3660, Cisco AS5300, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

This command sets the server location globally for all voice applications on the gateway.

For Nuance media servers that use the default installation, specify the URL as follows:

ivr asr-server rtsp://*host***:[***port***]/recognizer**

(*host* is the host name of the media server; **:***port* is optional.)

The location of the media server can be specified within a VoiceXML document, overriding the Cisco gateway configuration. For more information, refer to the *Cisco VoiceXML Programmer's Guide*.

Examples

The following example specifies that voice applications should use the ASR server named "asr_serv":

```
ivr asr-server rtsp://asr_serv/recognizer
```

Related Commands

Command	Description
ivr tts-server	Specifies the location of a media server that provides TTS functionality to voice applications.
ivr tts-voice-profile	Specifies the location of the voice profile that is used by the TTS server.

ivr autoload mode

To load files from TFTP to memory using either verbose or silent mode, use the **ivr autoload mode** command in global configuration mode. To disable this function, use the **no** form of this command.

ivr autoload mode {**verbose** [**url** *location*| **retry** *number*]} | {**silent** [**url** *location* | **retry** *number*]}

no ivr autoload mode

Syntax Description

verbose	Displays the file transfer activity to the console. This mode is recommended while debugging.
url *location*	URL that is used to locate the index file that contains a list of all available audio files.
retry *number*	(Optional) Number of times that the system tries to transfer a file when there are errors. This parameter applies to each file transfer. Range is from 1 to 5. Default is 3.
silent	Performs the file transfer in silent mode, meaning that no file transfer activity is displayed to the console.

Defaults

Silent

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

The index file contains a list of audio files (URL) that can be downloaded from the TFTP server. Use this command to download audio files from TFTP to memory. The command only starts up a background process. The background process (loader) does the actual downloading of the files.

The background process first reads the index file from either Flash or TFTP. It parses the files line by line looking for the URL. It ignores lines that start with # as comment lines. Once it has a correct URL, it tries to read that.au file into memory and creates a media object. If there are any errors during the reading of the file, it retries the configured number of times. If the mode is set to **verbose**, the loader logs the transaction to console. Once parsing has reached the end of the index file, the background process exits memory.

Perform the following checks before initiating the background process. If one of the checks fails, it indicates the background process is not started, and instead you see an error response to the command.

- Check if any prompt is being actively used (IVR is actively playing some prompts). If there are active prompts, the command fails, displaying the following error message (.au files are also referred to as prompts):

  ```
  command is not allowed when prompts are active
  ```

- Check if there is already a background process in progress. If there is a process, the command fails, displaying the following error:

  ```
  previous autoload command is still in progress
  ```

- Check if an earlier **ivr autoload url** command has already been configured. If an **ivr autoload url** command has already been configured, the user sees the following response when the command is issued:

  ```
  previous command is being replaced
  ```

- When the **no ivr autoload url** command is issued, if there was already an **ivr autoload url** command in progress, the original command is aborted.

The audio files (prompts) loaded using the **ivr autoload url** command are not dynamically swapped out of memory. They are considered to be autoloaded prompts, as opposed to dynamic prompts. (See the **ivr prompt memory** command for details on dynamic prompts.)

Examples

The following example configures verbose mode:

```
ivr autoload mode verbose url tftp://blue/orange/tclware/index4 retry 3
```

The following example shows the resulting index file:

```
more index4
tftp://blue/orange/tclware/au/en/en_one.au
tftp://blue/orange/tclware/au/ch/ch_one.au
tftp://blue/orange/tclware/au/ch/ch_one.au
```

The following example shows an index file on Flash memory:

```
flash:index
```

Related Commands

Command	Description
ivr prompt memory	Configures the maximum amount of memory that the dynamic audio files occupy in memory.

ivr autoload retry

To specify the number of times that the system tries to load audio files from TFTP to memory when there is an error, use the **ivr autoload retry** command in global configuration mode. To disable this function, use the **no** form of this command.

ivr autoload retry *number*

no ivr autoload retry

Syntax Description

number	Number of load attempts. Range is from 1 to 5. The default is 3.

Defaults

3 tries

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
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 system to try four times to load audio files:

```
ivr autoload retry 4
```

Related Commands

Command	Description
ivr prompt memory	Configures the maximum amount of memory that the dynamic audio files (prompts) occupy in memory.

ivr autoload url

To load files from a particular TFTP server (as indicated by a defined URL), use the **ivr autoload** command in global configuration mode. To disable this function, use the **no** form of this command.

ivr autoload url *location*

no ivr autoload url *location*

Syntax Description

url *location*	URL that is to be used to locate the index file that contains a list of all available audio files.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

The index file contains a list of audio files URLs that can be downloaded from the TFTP server. Use this command to download audio files from TFTP to memory. The command starts up a background process. The background process (loader) does the actual downloading of the files.

The background process first reads the index file from either Flash memory or TFTP. It parses the files line by line, looking for the URL. It ignores lines that start with # as comment lines. Once it has a correct URL, it tries to read that .au file into memory and creates a media object. If there are any errors during the reading of the file, it retries the configured number of times. If the *mode* is set to "verbose," in the ivr autoload mode command the loader logs the transaction to console. Once parsing has reached the end of the index file, the background process exits memory.

Perform the following checks before initiating the background process. If one of the checks fails, it indicates that the background process is not started, and instead you see an error response to the command.

- Check to see if any prompt is being actively used (IVR is actively playing some prompts). If there are active prompts, the command fails, displaying the following error message (.au files are also referred to as prompts):

```
command is not allowed when prompts are active
```

- Check to see if there is already a background process in progress. If there is a process, the command fails, displaying the following error:

```
previous autoload command is still in progress
```

- Check to see if an earlier **ivr autoload url** command has already been configured. If an **ivr autoload** command has already been configured, the user sees the following response when the command is issued:

  ```
  previous command is being replaced
  ```

- When the **no ivr autoload url** command is issued, If there is already an **ivr autoload url** command in progress, it is aborted.

The audio files (prompts) loaded using the **ivr autoload** command are not dynamically swapped out of memory. They are considered as autoloaded prompts as opposed to "dynamic" prompts. (See the **ivr prompt memory** command for details on dynamic prompts.)

Examples

The following example loads audio files from the TFTP server (located at //jurai/mgindi/tclware/index4):

```
ivr autoload url tftp://jurai/mgindi/tclware/index4
```

The following example shows the resulting index file:

```
more index4
tftp://jurai/mgindi/tclware/au/en/en_one.au
tftp://jurai/mgindi/tclware/au/ch/ch_one.au
tftp://jurai/mgindi/tclware/au/ch/ch_one.au
```

The following example shows an index file on Flash:

```
flash:index
```

Related Commands

Command	Description
ivr prompt memory	Configures the maximum amount of memory that the dynamic audio files (prompts) occupy in memory.

ivr prompt memory

To configure the maximum amount of memory that the dynamic audio files (prompts) occupy in memory, use the **ivr prompt memory** command in global configuration mode. To disable the maximum memory size, use the **no** form of this command.

ivr prompt memory *size* **files** *number*

no ivr prompt memory

Syntax Description

size	Maximum memory to be used by the free dynamic prompts, in kilobytes. Range is 128 to 16384. The default is 3000.
files *number*	Number of files that can stay in memory. Range is 50 to 1000. The default is 200.

Defaults

Memory size: 3000
Number of files: 200

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

When both the *number* and *size* parameters are specified, the minimum memory out of the two is used for memory calculations.

All the prompts that are not autoloaded or fixed are considered dynamic. Dynamic prompts are loaded in to memory from TFTP or Flash, as and when they are needed. When they are actively used for playing prompts, they are considered to be in "active" state. However, once the prompt playing is complete, these prompts are no longer active and are considered to be in a free state.

The free prompts either stay in memory or are removed from memory depending on the availability of space in memory for these free prompts. This command essentially specifies a maximum memory to be used for these free prompts.

The free prompts are saved in memory and are queued in a wait queue. When the wait queueis full (either because the totally memory occupied by the free prompts exceeds the maximum configured value or the number of files in the wait queue exceeds maximum configured), oldest free prompts are removed from memory.

Examples

The following example sets memory size to 2048 KB and number of files to 500:

```
ivr prompt memory 2048 files 500
```

Related Commands

Command	Description
ivr autoload	Loads files from a particular TFTP server.
show call prompt-mem-usage	Displays the memory site use by prompts.

ivr prompt streamed

To stream audio prompts from particular media types during playback, use the **ivr prompt streamed** command in global configuration mode. To reset to the default, use the **no** form of this command.

ivr prompt streamed {**all** | **flash** | **http** | **none** | **tftp**}

no ivr prompt streamed {**all** | **flash** | **http** | **none** | **tftp**}

Syntax Description

all	All audio prompts, from all URL types (Flash memory, HTTP, TFTP).
flash	Audio prompts from Flash memory.
http	Audio prompts from an HTTP URL. This is the default value.
none	No audio prompts from any media type.
tftp	Audio prompts from a TFTP URL.

Defaults

Only audio prompts from HTTP URLs are streamed during playback; other media types are not streamed.

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the following platforms: Cisco 3640, Cisco 3660, Cisco AS5300, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

If this command is not entered, audio prompts from HTTP servers are streamed; audio prompts from Flash and TFTP servers are not streamed. To enable streaming for multiple media types, either enter this command for each URL type or enter the **ivr prompt streamed all** command.

Note Prompts from a Real Time Streaming Protocol (RTSP) server are always streamed during playback.

Examples

The following example indicates that audio prompts from Flash memory and TFTP are streamed when they are played back:

```
ivr prompt streamed flash
ivr prompt streamed tftp
```

Related Commands

Command	Description
ivr prompt memory	Sets the maximum amount of memory that dynamic audio prompts can occupy in memory.

ivr record memory session

To set the maximum amount of memory that can be used to record voice messages during a single call session, use the **ivr record memory session** command in global configuration mode. To reset to the default, use the **no** form of this command.

ivr record memory session *kilobytes*

no ivr record memory session

Syntax Description

kilobytes	Memory size, in kilobytes. Range is 0 to 256000. The default is 256.

Defaults

256 KB

Command Modes

Global configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco AS5300.
12.2(11)T	This command was implemented on the following platforms: Cisco 3640, Cisco 3660, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

Use this command to limit the maximum memory allowed for audio recordings during a single call session on a VoiceXML-enabled gateway.

Note This command configures memory limits only for voice messages recorded to local memory on the gateway. Memory limits are not configurable on the gateway for HTTP, Real Time Streaming Protocol (RTSP), or Simple Mail Transfer Protocol (SMTP) recordings.

Examples

The following example sets the maximum memory limit to 512 KB for a single call session:

```
ivr record memory session 512
```

Related Commands

Command	Description
ivr record memory system	Sets the maximum amount of memory that can be used to store all voice recordings on the VoiceXML-enabled gateway.

ivr record memory system

To set the maximum amount of memory that can be used to store all voice recordings on the gateway, use the **ivr record memory system** command in global configuration mode. To reset to the default, use the **no** form of this command.

ivr record memory system *kilobytes*

no ivr record memory system

Syntax Description

kilobytes	Memory limit, in kilobytes. Range is 0 to 256000. If 0 is configured, the RAM recording function is disabled on the gateway. The default for Cisco 3640 and Cisco AS5300 is 10000. The default for Cisco 3660, Cisco AS5350, and Cisco AS5400 is 20000.

Defaults

Cisco 3640 and Cisco AS5300: 10,000 KB
Cisco 3660, Cisco AS5350, and Cisco AS5400: 20,000 KB

Command Modes

Global configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco AS5300.
12.2(11)T	This command was implemented on the following platforms: Cisco 3640, Cisco 3660, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

Use this command to limit the maximum amount of gateway memory that is used for storing all voice recordings.

Note This command configures memory limits only for voice messages recorded to local memory on the gateway. Memory limits are not configurable on the gateway for HTTP, Real Time Streaming Protocol (RTSP), or Simple Mail Transfer Protocol (SMTP) recordings.

Examples

The following example sets the total memory limit for all recordings to 8000 KB:

```
ivr record memory system 8000
```

Related Commands

Command	Description
ivr record memory session	Sets the maximum amount of memory that can be used to record voice messages during a single call session.

ivr tts-server

To specify the location of an external media server that provides text-to-speech (TTS) functionality to voice applications, use the **ivr tts-server** command in global configuration mode. To remove the server location, use the **no** form of this command.

ivr tts-server *url*

no ivr tts-server

Syntax Description

url	Location of the TTS resource on the media server, in URL format.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the following platforms: Cisco 3640, Cisco 3660, Cisco AS5300, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

This command sets the server location globally for all voice applications on the gateway.

For Nuance media servers that use the default installation, specify the URL as follows:

ivr tts-server rtsp://*host***:***port***/synthesizer**

(*host* is the host name of the media server; **:***port* is optional.)

The location of the media server can be specified within a VoiceXML document, overriding the Cisco gateway configuration. For more information, refer to the *Cisco VoiceXML Programmer's Guide*.

To specify the voice profile that the TTS server uses for voice synthesis operations, use the **ivr tts-voice-profile** command.

Examples

The following example specifies that voice applications should use the TTS server named "tts_serv":

```
ivr tts-server rtsp://tts_serv/synthesizer
```

Related Commands

Command	Description
ivr asr-server	Specifies the location of a media server that provides ASR functionality to IVR applications.
ivr tts-voice-profile	Specifies the location of the voice profile that is used by the TTS server.

ivr tts-voice-profile

To specify the location of the voice profile that is used by text-to-speech (TTS) servers, use the **ivr tts-voice-profile** command in global configuration mode. To remove the voice profile, use the **no** form of this command.

ivr tts-voice-profile *url*

no ivr tts-voice-profile

Syntax Description

url	Location of the TTS voice profile file, in URL format.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the following platforms: Cisco 3640, Cisco 3660, Cisco AS5300, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

This command specifies the voice profile that a TTS server uses for voice synthesis operations. The voice profile is a W3C Simple Markup Language (SML) file that specifies voice parameters like gender, speed, and so forth. The TTS server uses this voice profile unless the markup file that it is translating has overriding values.

The TTS voice profile can be stored on an HTTP server or on RTSP, TFTP, or FTP servers if the media sever supports these locations.

The TTS voice profile location can also be specified in the VoiceXML document by using the Cisco proprietary property com.cisco.tts-voice-profile. The VoiceXML property in the document overrides the value that is configured by using this command.

To specify the location of the external media server that is providing TTS functionality, use the **ivr tts-server** command.

Examples

The following example tells the TTS server to use the voice profile file named "vprofil2", which is located on an HTTP server:

```
ivr tts-voice-profile http://ttserver/vprofil2.sml
```

Related Commands

Command	Description
ivr asr-server	Specifies the location of a media server that provides ASR functionality to IVR applications.
ivr tts-server	Specifies the media server that provides TTS functionality to IVR applications.