Cisco IOS Voice Commands: V through W

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*.

vad (dial peer)

To enable voice activity detection (VAD) for the calls using a particular dial peer, use the **vad** command in dial-peer configuration mode. To disable VAD, use the **no** form of this command.

vad [aggressive]

no vad [aggressive]

Syntax Description

aggressive	Reduces noise threshold from –78 dBm to –62 dBm. Available only when session protocol multicast is configured.

Defaults

VAD is enabled.

Aggressive VAD is enabled in multicast dial peers.

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)T	This command was implemented as a dial-peer command on the Cisco MC3810 (in prior releases, the **vad** command was available only as a voice-port command).
12.2(11)T	The **aggressive** keyword was added.

Usage Guidelines

Use this command to enable voice activity detection. With VAD, voice data packets fall into three categories: speech, silence, and unknown. Speech and unknown packets are sent over the network; silence packets are discarded. The sound quality is slightly degraded with VAD, but the connection monopolizes much less bandwidth. If you use the **no** form of this command, VAD is disabled and voice data is continuously sent to the IP backbone. When configuring voice gateways to handle fax calls, VAD should be disabled at both ends of the IP network because it can interfere with the successful reception of fax traffic.

When the **aggressive** keyword is used, the VAD noise threshold is reduced from –78 to –62 dBm. Noise that falls below the –62 dBm threshold is considered to be silence and is not sent over the network. Additionally, unknown packets are considered to be silence and are discarded.

On the Cisco MC3810, VAD can also be assigned to the voice port using the **vad (voice-port)** command. On the Cisco MC3810 multiservice concentrator, if you enable VAD on the dial peer for Voice over Frame Relay switched calls or permanent calls, the dial-peer setting overrides the VAD setting on the voice port.

Note On the Cisco MC3810, the **vad (dial-peer)** command is enabled by default. The **vad (voice-port)** command is disabled by default.

Examples

The following example enables VAD for a Voice over IP (VoIP) dial peer, starting from global configuration mode:

```
dial-peer voice 200 voip
 vad
```

Related Commands

Command	Description
comfort-noise	Generates background noise to fill silent gaps during calls if VAD is activated.
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
vad (voice-port)	Enables VAD for the calls using a particular voice port.

vad (voice-port)

To enable voice-activity detection (VAD) for the calls using a particular voice port, use the **vad** command in voice-port configuration mode. To disable VAD, use the **no** form of this command.

vad

no vad

Syntax Description

This command has no arguments or keywords.

Defaults

VAD is not enabled.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced as a voice-port command on the Cisco MC3810.

Usage Guidelines

This command applies to Voice over Frame Relay and Voice over ATM on Cisco MC3810 multiservice concentrators.

Use this command to enable voice activity detection. With VAD, silence is not sent over the network; only audible speech is sent. If you enable VAD, the sound quality is slightly degraded but the connection monopolizes much less bandwidth. If you use the **no** form of this command, VAD is disabled on the voice port. When configuring voice gateways to handle fax calls, VAD should be disabled at both ends of the IP network because it can interfere with the successful reception of fax traffic.

Note It is recommended that you use the **vad** command in dial-peer configuration mode.

Examples

The following example enables VAD:

```
voice-port 1/1
 vad
```

Related Commands

Command	Description
comfort-noise	Generates background noise to fill silent gaps during calls if VAD is activated.
vad (dial peer)	Enables VAD for the calls using a particular dial peer.

vbd-playout-delay maximum

To enable maximum ATM adaptation layer 2 (AAL2) voice-band-detection playout-delay buffer on a Cisco router, use the **vbd-playout-delay** command in voice-service configuration mode. To reset to the default, use the **no** form of this command.

vbd-playout-delay maximum *time*

no vbd-playout-delay maximum

Syntax Description

time	Playout delay, in milliseconds. Range is from 40 to 1700. Default is 200.

Defaults

200 milliseconds

Command Modes

Voice-service configuration

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco 2600 series and Cisco 3660.

Examples

The following example sets the AAL2 voice-band-detection playout-buffer delay to a maximum of 202 milliseconds:

```
voice service voatm
 session protocol aal2
 vbd-playout-delay maximum 202
```

Related Commands

Command	Description
voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

vbd-playout-delay minimum

To enable minimum ATM adaptation layer 2 (AAL2) voice-band-detection playout-delay buffer on a Cisco router, use the **vbd-playout-delay** command in voice-service configuration mode. To reset to the default, use the **no** form of this command.

vbd-playout-delay minimum *time*

no vbd-playout-delay minimum

Syntax Description

time	Playout delay, in milliseconds. Range is from 4 to 1700. Default is 4.

Defaults

4 milliseconds

Command Modes

Voice-service configuration

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco 2600 series and Cisco 3660.

Examples

The following example sets the AAL2 voice-band-detection playout-buffer delay to a minimum of 6 milliseconds:

```
voice service voatm
 session protocol aal2
 vbd-playout-delay minimum 6
```

Related Commands

Command	Description
voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

vbd-playout-delay mode

To configure voice-band-detection playout-delay adaptation mode on a Cisco router, use the **vbd-playout-delay** command in voice-service configuration mode. To disable this mode, use the **no** form of this command.

vbd-playout-delay mode [**fixed** | **passthrough**]

no vbd-playout-delay mode [**fixed** | **passthrough**]

Syntax Description

fixed	Sets jitter buffer to a constant delay, in milliseconds.
passthrough	Sets jitter buffer passthrough to DRAIN_FILL for clock compensation.

Defaults

Voice-band-detection playout-delay adaptation mode is disabled.

Command Modes

Voice-service configuration

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco 2600 series and Cisco 3660.

Usage Guidelines

Use this command to set the playout jitter buffer. When a voice band is detected, the call uses G.711 codec, and the playout delay values that you set are picked up. The original voice-call parameters are restored after the fax or modem call is completed.

Examples

The following example configures ATM adaptation layer 2 (AAL2) voice-band-detection playout-delay adaptation mode and sets the mode to fixed:

```
voice service voatm
 session protocol aal2
 vbd-playout-delay mode fixed
```

Related Commands

Command	Description
voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

vbd-playout-delay nominal

To enable nominal ATM adaptation layer 2 (AAL2) voice-band-detection playout-delay buffer on a Cisco router, use the **vbd-playout-delay** command in voice-service configuration mode. To reset to the default, use the **no** form of this command.

vbd-playout-delay nominal *time*

no vbd-playout-delay nominal

Syntax Description

time	Playout delay, in milliseconds. Range is from 0 to 1500. Default is 100.

Defaults

100 milliseconds

Command Modes

Voice-service configuration

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco 2600 series and Cisco 3660.

Examples

The following example sets the nominal AAL2 voice-band-detection playout-delay buffer to 202 milliseconds:

```
voice service voatm
 session protocol aal2
 vbd-playout-delay nominal 202
```

Related Commands

Command	Description
voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

vbr-rt

To configure the real-time variable bit rate (VBR) for VoATM voice connections, use the **vbr-rt** command in the appropriate configuration mode. To disable VBR for voice connections, use the **no** form of this command.

vbr-rt *peak-rate average-rate burst*

no vbr-rt

Syntax Description

peak-rate	Peak information rate (PIR) for the voice connection, in kbps. If it does not exceed your carrier's line rate, set it to the line rate. Range is from 56 to 10000.
average-rate	Average information rate (AIR) for the voice connection, in kbps.
burst	Burst size, in number of cells. Range is from 0 to 65536.

Defaults

No real-time VBR settings are configured.

Command Modes

For an ATM permanent virtual connection (PVC) or switched virtual circuit (SVC): Interface-ATM-VC configuration

For a virtual circuit (VC) class: VC-class configuration

For ATM VC bundle members: Bundle-vc configuration

Command History

Release	Modification
12.0	This command was introduced on the Cisco MC3810.
12.1(5)XM	This command was implemented on the Cisco 3600 series and modified to support SGCP and MGCP.
12.2(2)T	This command was integrated into this release.
12.2(11)T	This command was implemented on the Cisco AS5300 and Cisco AS5850.

Usage Guidelines

This command configures traffic shaping between voice and data PVCs. Traffic shaping is required so that the carrier does not discard calls. To configure voice and data traffic shaping, you must configure the peak, average, and burst options for voice traffic. Configure the burst value if the PVC will carry bursty traffic. Peak, average, and burst values are needed so that the PVC can effectively handle the bandwidth for the number of voice calls.

Calculate the minimum peak, average, and burst values for the number of voice calls as follows:

Peak Value

Peak value = (2 x the maximum number of calls) x 16K = _______________

Average Value

Calculate according to the maximum number of calls that the PVC will carry times the bandwidth per call. The following formulas give you the average rate in kbps:

- For VoIP:
  - G.711 with 40- or 80-byte sample size:

    Average value = max calls x 128K = _______________
  - G.726 with 40-byte sample size:

    Average value = max calls x 85K = _______________
  - G.729a with 10-byte sample size:

    Average value = max calls x 85K = _______________
- For VoATM adaptation layer 2 (VoAAL2):
  - G.711 with 40-byte sample size:

    Average value = max calls x 85K = _______________
  - G.726 with 40-byte sample size:

    Average value = max calls x 43K = _______________
  - G.729a with 10-byte sample size:

    Average value = max calls x 43K = _______________

If voice activity detection (VAD) is enabled, bandwidth usage is reduced by as much as 12 percent with the maximum number of calls in progress. With fewer calls in progress, bandwidth savings are less.

Burst Value

Set the burst size as large as possible, and never less than the minimum burst size. Guidelines are as follows:

- Minimum burst size = 4 x number of voice calls = _______________
- Maximum burst size = maximum allowed by the carrier = _______________

When you configure data PVCs that will be traffic shaped with voice PVCs, use aal5snap encapsulation and calculate the overhead as 1.13 times the voice rate.

Examples

The following example configures the traffic-shaping rate for ATM PVC 20. Peak, average, and burst rates are calculated based on a maximum of 20 calls on the PVC.

```
pvc 20
 encapsulation aal5mux voice
 vbr-rt 640 320 80
```

Related Commands

Command	Description
encapsulation aal5	Configures the AAL and encapsulation type for an ATM PVC, SVC, or VC class.

vcci

To identify a permanent virtual circuit (PVC) to the call agent, use the **vcci** command in ATM virtual circuit (VC) configuration mode. To restore the default value, use the **no** form of this command.

vcci *pvc-identifier*

no vcci

Syntax Description

pvc-identifier	Identifier for the PVC. Range is from 0 to 32767. There is no default.

Defaults

No default behavior or values

Command Modes

ATM virtual circuit configuration mode

Command History

Release	Modification
12.1(5)XM	This command was introduced.
12.2(2)T	This command was integrated into this release.
12.2(11)T	This command was implemented on the Cisco AS5300 and Cisco AS5850.

Usage Guidelines

The *pvc-identifier* argument is a unique 15-bit value for each PVC. The call agent sets up a call with the gateway by specifying the PVC using the *pvc-identifier*.

Examples

The following example shows how to assign a PVC identifier:

```
Router(config-if-atm-vc)# vcci 5278
```

Related Commands

Command	Description
mgcp	Starts the MGCP daemon.
pvc	Creates an ATM PVC for voice traffic.

vm-device-id (ephone)

To define the voice-mail ID string, use the **vm-device-id** command in ephone configuration mode. To disable this feature, use the **no** form of this command.

vm-device-id *id-string*

no command *id-string*

Syntax Description

id-string	Voice-mail-device port identification (ID) string; for example, CiscoUM-VI1 for the first port and CiscoUM-VI2 for the second port.

Defaults

No default behavior or values

Command Modes

Ephone configuration

Command History

Release	Modification
12.2(2)XT	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600, Cisco 3600, and Cisco IAD2420.
12.2(8)T	This command was 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

Use this command to define the voice-mail-device ID string. The voice-mail port registers with a device ID instead of a MAC address. To distinguish among different voice-mail ports, voice-mail-device ID is used. The voice-mail-device ID is configured to a Cisco IP phone port, which maps to a corresponding voice-mail port.

Examples

The following example shows how to set the voice-mail device ID to CiscoUM-V11:

```
Router(config) ephone 1
Router(config-ephone) vm-device ID CiscoUM-VI1
```

Related Commands

Command	Description
voicemail (telephony-service)	Configures the telephone number that is speed-dialed when the messages button on a Cisco IP phone is pressed.

vm-integration

To enable voice-mail integration with dual-tone multifrequency (DTMF) and analog voice-mail systems and to enter voice-mail integration configuration mode, use the **vm-integration** command in global configuration mode. To disable voice-mail integration, use the **no** form of this command.

vm-integration

no vm-integration

Syntax Description

This command has no arguments or keywords.

Defaults

Voice-mail integration is disabled.

Command Modes

Global configuration

Command History

Release	Modification
12.2(2)XT	For Cisco IOS Telephony Service, this command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	For Cisco IOS Telephony Service, this command was implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	For Cisco IOS Telephony Service,this command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	For Cisco IOS Telephony Service, this command was implemented on the Cisco 1760.
12.2(13)T	This command was implemented on Cisco Survivable Remote Site Telephony, Version 2.02.

Usage Guidelines

The **vm-integration** command allows you to enter voice-mail integration configuration mode and allows integration with DTMF and analog voice-mail systems.

Examples

The following example enters voice-mail integration configuration mode:

```
Router(config) vm-integration
Router(config-vm-integration)
```

Related Commands

Command	Description
pattern direct	Configures the DTMF pattern for direct dialing when the user presses the messages button on the phone to access voice-mail messages.
pattern ext-to-ext busy	Configures the DTMF pattern for forward dialing when an internal extension calls another busy extension and the call is forwarded to a voice-mail system.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension that does not answer and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF pattern for forward dialing when an external trunk call reaches a busy extension and the call is forwarded to a voice-mail system.
pattern trunk-to-ext no-answer	Configures the DTMF pattern for forward dialing when an external trunk call reaches another extension and the call is forwarded to a voice-mail system.

vofr

To enable Voice over Frame Relay (VoFR) on a specific data-link connection identifier (DLCI) and to configure specific subchannels on that DLCI, use the **vofr** command in frame relay DLCI configuration mode. To disable VoFR on a specific DLCI, use the **no** form of this command.

Switched Calls

vofr [**data** *cid*] [**call-control** [*cid*]]

no vofr [**data** *cid*] [**call-control** [*cid*]]

Switched Calls to Cisco MC3810 Multiservice Concentrators Running Cisco IOS Releases Before 12.0(7)XK and 12.1(2)T

vofr [**cisco**]

no vofr [**cisco**]

Cisco-Trunk Permanent Calls

vofr data *cid* **call-control** *cid*

no vofr data *cid* **call-control** *cid*

Cisco-Trunk Permanent Calls to Cisco MC3810 Multiservice Concentrators Running Cisco IOS Releases Before 12.0(7)XK and 12.1(2)T

vofr cisco

no vofr cisco

FRF.11 Trunk Calls

vofr [**data** *cid*] [**call-control** *cid*]

no vofr [**data** *cid*] [**call-control** *cid*]

Syntax Description

data *cid*	(Required for Cisco-trunk permanent calls; optional for switched calls) Reserved subchannel for data other than the default subchannel. Range is from 4 to 255. Default is 4.
call-control *cid*	(Optional) Reserved subchannel for call-control signaling. Range is from 4 to 255. Default is 5. Not supported on the Cisco MC3810.
cisco *cid*	(Optional) Reserved subchannel for Cisco-proprietary voice encapsulation for VoFR. Data is carried on CID 4 and call-control on CID 5. This option is required when configuring switched calls or Cisco trunks to Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK and 12.1(2)T. If you are configuring switched calls or Cisco trunks to Cisco MC3810 running Cisco IOS Release 12.0(7)XK and 12.1(2)T and later releases, do not use this option.

Defaults

Disabled

Command Modes

Frame relay DLCI configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, and Cisco MC3810.
12.0(4)T	This command was integrated into this release.
12.0(7)XK	The use of the **cisco** option was modified. Beginning in this release, use the **cisco** option only when configuring connections to Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK and 12.1(2)T.
12.1(2)T	This command was integrated into this release.

Usage Guidelines

Table 159 lists the different options of the **vofr** command and which combination of options is used beginning in Cisco IOS Release 12.0(7)XK and Release 12.1(2)T.

Table 159 Combinations of the vofr Command

Type of Call	Command Combination to Use
Switched call (user dialed or auto-ringdown) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** [*cid*]][1]
Switched call (user dialed or auto-ringdown) to a Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK and 12.1(2)T	**vofr cisco**[2]
Cisco-trunk permanent call (private-line) to other routers supporting VoFR	**vofr data** *cid* **call-control** *cid*
Cisco-trunk permanent call (private-line) to a Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK and 12.1(2)T	**vofr cisco**
FRF.11 trunk call (private-line) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** *cid*][3]

1. The recommended form of this command to use is **vofr data 4 call-control 5**.
2. This command consumes data CID 4 and call-control CID 5.
3. For FRF.11 trunk calls, the call-control option is not required. It is required only if you mix FRF.11 trunk calls with other types of voice calls on the same PVC.

Usage Restrictions for Cisco IOS Releases Before 12.0(7)XK and 12.1(2)T

This section describes restrictions for using the **vofr** command in releases before Cisco IOS Release 12.0(7)XK and 12.1(2)T. Beginning in Cisco IOS Release 12.0(7)XK and 12.1(2)T, these restrictions no longer apply.

When you use the **vofr** command without the **cisco** option, all subchannels on the DLCI are configured for FRF.11 encapsulation. If you enter the **vofr** command without any keywords or arguments, the data subchannel is CID 4 and there is no call-control subchannel.

Table 160 describes special conditions and restrictions for the use of the **vofr** command on the Cisco MC3810 running releases before 12.0(7)XK and 12.1(2)T.

Table 160 Using the vofr Command with the Cisco MC3810

Type of Call	Conditions and Restrictions
FRF.11 trunks	1. Do *not* use the **cisco** option or the **call-control** option. 2. Use **vofr** or **vofr data** *cid*.
Cisco trunks	1. Must use **vofr cisco**.
switched-vofr	1. Must use **vofr cisco**.

If you select the "data" option, enter a numeric value to complete the command. If you select the **call-control** option, you do not enter a numeric value if you wish to accept the default call-control subchannel. See the following examples for clarification.

When you use the **vofr** command on a Cisco MC3810 multiservice concentrator without the "cisco" option, switched calls are not permitted. You can make only permanent FRF.11-trunk calls.

Note It is not possible to configure the **call-control** option on a Cisco MC3810. If you configure this option, the setting is ignored.

Examples

The following example, beginning in global configuration mode, shows how to enable VoFR on serial interface 1/1, DLCI 100 on a Cisco 2600 series, Cisco 3600 series, or Cisco 7200 series router or on a Cisco MC3810. The example configures CID 4 for data; no call-control CID is defined.

```
interface serial 1/1
 frame-relay interface-dlci 100
 vofr
```

To configure CID 4 for data and CID 5 for call-control (both defaults), enter the following command:

```
vofr call-control
```

To configure CID 10 for data and CID 15 for call-control, enter the following command:

```
vofr data 10 call-control 15
```

To configure CID 4 for data and CID 15 for call-control, enter the following command:

```
vofr call-control 15
```

To configure CID 10 for data and CID 5 for call-control, enter the following command:

```
vofr data 10 call-control
```

To configure CID 10 for data with no call-control, enter the following command:

```
vofr data 10
```

To configure a Cisco router or Cisco MC3810 for a VoFR application with an older release of the Cisco MC3810 (before Release 12.0(3)XG), enter the following command:

```
vofr cisco
```

Related Commands

Command	Description
class	Assigns a VC class to a PVC.
frame-relay interface-dlci	Assigns a DLCI to a specified Frame Relay subinterface.

voice

To enable voice resource pool services for resource pool management, use the **voice** command in service profile configuration mode. To disable voice services, use the **no** form of this command.

voice

no voice

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Service profile configuration mode

Command History

Release	Modification
12.2(2)XA	This command was introduced on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into this release.

Examples

The following example shows that voice service is available and enables voice resource pool service using the **voice** command in service profile configuration mode:

```
Router(config)# resource-pool profile service voip

Router(config-service-profile)# ?
  Service Profile Configuration Commands:
  default    Set a command to its defaults
  exit       Exit from resource-manager configuration mode
  help       Description of the interactive help system
  modem      Configure modem service parameters
  no         Negate a command or set in its defaults
  voice      Configure voice service parameters

Router(config-service-profile)# voice
```

Related Commands

Command	Description
resource-pool enable	Enables resource pool management.
resource-pool profile service voip	Defines the VoIP service profile for resource pool management.

voice call capacity mir

To set the value for the minimum interval between reporting (MIR), use the **voice call capacity mir** command in global configuration mode. To turn off these attributes, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **capacity mir** *seconds*

no voice call {**carrier** | **trunk-group** | **prefix**} **capacity mir**

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
seconds	Minimum interval, in seconds. Range is from 1 to 3600. Default is 10. This value cannot be set higher than the time configured for the **capacity update interval**.

Defaults

10 seconds.

Command Modes

Global configuration.

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

Because the available circuit (AC) attribute of a destination is very dynamic, reporting of this attribute should be handled carefully. AC should be reported as frequently as possible so that the location server has better information about the resources. However, the location server should not be overwhelmed with too many updates.

All of the AC reporting, called the *interesting point of AC*, is performed when the specified event happens within the *minimum interval between reporting* (MIR) time since last reporting. This command sets the amount of time used for the interval to control the number of interesting points that are reported so not to overwhelm the location server with too many AC updates.

The *seconds* argument cannot be set higher than the time configured for the **capacity update interval**.

Examples

The following example shows the minimum interval between reporting for the carrier address family set to 25 seconds:

```
Router(config)# voice call carrier capacity mir 25
```

Related Commands

Command	Description
capacity update interval (dial peer)	Changes the capacity update for prefixes associated with a dial peer.
capacity update interval (trunk group)	Change the capacity update for carriers or trunk groups.
voice call capacity stw	Set the value for STW.

voice call capacity stw

To set the value for smoothing transition time for weight (STW), use the **voice call capacity stw** command in global configuration mode. To turn off these attributes, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **capacity stw** *seconds*

no voice call {**carrier** | **trunk-group** | **prefix**} **capacity stw**

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
seconds	Transition time, in seconds. Range is from 0 to 60. Default is 10.

Defaults

10 seconds.

Command Modes

Global configuration.

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

Because the available circuit (AC) attribute of a destination is very dynamic, reporting of this attribute should be handled carefully. AC should be reported as frequently as possible so that the location server has better information about the resources. However, the location server should not be overwhelmed with too many updates.

A smoothing algorithm is applied to the quantity of AC being reported. This algorithm eliminates reporting of noise. The degree of smoothing can be configured with the **voice call capacity stw** command. This command sets the smoothing transition time for weight, which is the time it takes for current smoothed value of AC to come half way between the current smoothed value and the current instantaneous value of AC. Lower **stw** values speed the smoothed value of AC as it approaches the instantaneous value of AC. When **stw** is set to 0, the smoothed value is always equal to the instantaneous value of AC.

Examples

The following example shows the smoothing time for weight for the carrier address family set to 25 seconds:

```
Router(config)# voice call carrier capacity stw 25
```

Related Commands

Command	Description
capacity update interval (dial peer)	Changes the capacity update for prefixes associated with a dial peer.
capacity update interval (trunk group)	Change the capacity update for carriers or trunk groups.
voice call capacity mir	Set the value for MIR.

voice call capacity reporting

To turn on the reporting of maxima (first derivative) or inflection (second derivative) points in available capacity, use the **voice call capacity reporting** command in global configuration mode. To turn off the reporting, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **capacity reporting** {**maxima** | **inflection**}

no voice call {**carrier** | **trunk-group** | **prefix**} **capacity reporting** {**maxima** | **inflection**}

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
maxima	Maxima (first derivative) point in available capacity.
inflection	Inflection (second derivative) point in available capacity.

Defaults

Capacity reporting is turned off.

Command Modes

Global configuration.

Command History

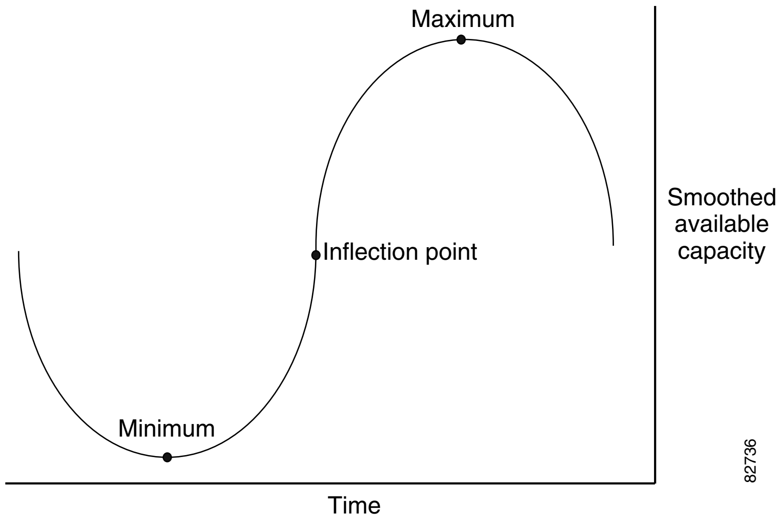
Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

The smoothed curve of the available circuits (AC) has maxima, minima, and inflection points. When the curve has reached these points, this represents a change in the call rate.

Maximum, minimum and inflection points are illustrated in Figure 5.

Figure 5 Maximum, Minimum, and Inflection Points for Available Capacity



Examples

The following example shows the reporting of the available capacity inflection point on the trunk group is turned on:

```
Router(config)# voice call trunk-group capacity reporting inflection
```

Related Commands

Command	Description
voice call capacity mir	Sets the values for the minimum interval between reporting (MIR) and smoothing transition time for weight (STW).
voice call capacity timer interval	Sets the periodic interval for reporting capacity from carrier, trunk group, or prefix databases
voice call trigger hwm	Sets the value for percentage change, low water mark and high water mark in the available capacity in the trunk group or prefix databases.

voice call capacity timer interval

To set the periodic interval for reporting capacity from carrier, trunk group, or prefix databases, use the **voice call capacity timer interval** command in global configuration mode. To turn off the interval, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **capacity timer interval** *seconds*

no voice call {**carrier** | **trunk-group** | **prefix**} **capacity timer interval** *seconds*

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
seconds	Interval, in seconds. Range is from 10 to 3600. Default is 25.

Defaults

25 seconds

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

For the reporting interval, a periodic timer called the capacity update timer handles updates of available circuit (AC) information and can be configured using the **voice call capacity timer interval** command. For example, if AC has changed since the last reporting, the AC is again reported when the capacity update timer expires.

Examples

The following example sets the timer interval for the prefixes set at 15 seconds:

```
Router(config)# voice call prefix capacity timer interval 15
```

Related Commands

Command	Description
voice call capacity mir	Sets the values for the MIR and STW.
voice call capacity reporting	Turns on the reporting of maxima (first derivative) or inflection (second derivative) points in available capacity.
voice call trigger hwm	Sets the value for percentage change, low water mark and high water mark in the available capacity in the trunk group or prefix databases.

voice call convert-discpi-to-prog

To convert a disconnect message with a progress indicator (PI) to a progress message, use the **voice call convert-discpi-to-prog** command in global configuration mode. To return to the default condition, use the **no** form of this command.

voice call convert-discpi-to-prog [**tunnel-IEs** | **always** [**tunnel-IEs**]]

no voice call convert-discpi-to-prog

Syntax Description

tunnel-IEs	(Optional) Information elements (IEs) are carried in the progress message.
always	(Optional) Converts disconnect message with a PI to a progress message in both preconnected and connected states.

Defaults

A disconnect message with a PI is not converted to a progress message.

Command Modes

Global configuration

Command History

Release	Modification
12.2(1)	This command was introduced.
12.3(6)	The **tunnel-1Es** keyword was added.
12.3(4)XQ	The **always** keyword with the **tunnel-IEs keyword** were added.
12.3(8)T	The **always** keyword with the **tunnel-IEs keyword** were added.
12.3(9)	The **always** keyword with the **tunnel-1Es keyword** were added.

Usage Guidelines

The **voice call convert-discpi-to-prog** command turns an ISDN disconnect message into a progress message. If you use the **tunnel-IEs** keyword, the information elements are not dropped when the disconnect message is converted to a progress message.

Examples

The following example changes a disconnect with PI to a progress message containing information elements (IEs):

```
voice call convert-discpi-to-prog tunnel-IEs
```

The following example changes a disconnect with PI to a progress message in the preconnected and connected states:

```
voice call convert-discpi-to-prog always
```

Related Commands

Command	Description
disc_pi_off	Enables an H.323 gateway to disconnect a call when it receives a disconnect message with a PI.

voice call csr data-points

To set the number of call-success-rate (CSR) data points, use the **voice call csr data-points** command in global configuration mode. To disable the setting, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **csr data-points** *value*

no voice call {**carrier** | **trunk-group** | **prefix**} **csr data-points** *value*

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
value	Number of data-points. Range is from 10 to 50. Default is 30.

Defaults

30 data points

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Examples

The following example sets the CSR data points for trunk groups at 10:

```
Router(config)# voice call trunk-group csr data-points 10
```

Related Commands

Command	Description
voice call csr recording interval	Sets the recording interval for CSR.
voice call csr reporting interval	Sets the reporting interval for CSR.

voice call csr recording interval

To set the recording interval for call success rates (CSR), use the **voice call csr recording interval** command in global configuration mode. To disable the interval, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **csr recording interval** *minutes*

no voice call {**carrier** | **trunk-group** | **prefix**} **csr recording interval** *minutes*

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
minutes	Recording interval, in minutes. Range is from 10 to 1000. Default is 60.

Defaults

60 minutes

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Examples

The following example sets the CSR recording interval for prefixes at 30 minutes:

```
Router(config)# voice call carrier csr recording interval 30
```

Related Commands

Command	Description
voice call csr data-points	Sets the number of call success rate (CSR) data points.
voice call csr reporting interval	Sets the reporting interval for CSR.

voice call csr reporting interval

To set the reporting interval for call success rate (CSR), use the **voice call csr reporting interval** command in global configuration mode. To disable the CSR recording interval, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **csr reporting interval** *seconds*

no voice call {**carrier** | **trunk-group** | **prefix**} **csr reporting interval** *seconds*

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
seconds	Reporting interval, in seconds. Range is from 10 to 10000. Default is 25.

Defaults

25 seconds

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Examples

The following example sets the CSR reporting interval for trunk groups at 40 seconds:

```
Router(config)# voice call carrier csr reporting interval 40
```

Related Commands

Command	Description
voice call csr data-points	Sets the number of CSR data points.
voice call csr recording interval	Sets the recording interval for CSR.

voice call debug

To debug a voice call, use the **voice call debug** command in global configuration mode. To display a full globally unique identifier (GUID) or header as explained in the Usage Guidelines section, use the **no** form of this command.

voice call debug full-guid | short-header

no voice call debug full-guid | short-header

Syntax Description

full-guid	Displays the GUID in a 16-byte header. **Note** When you use the **no** version of this command with the **full-guid** keyword, the short 6-byte version displays. This is the default.
short-header	Displays the CallEntry ID in the header without displaying the GUID or module-specific parameters.

Defaults

The short 6-byte header displays.

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	The new debug header was added to the following: Cisco 2600 series, Cisco 3620, Cisco 3640, Cisco 3660 series, Cisco AS5350, Cisco AS5400, Cisco AS5850, Cisco AS5300, Cisco AS5800, and Cisco MC3810.
12.2(15)T	The header-only argument was removed and the short-header argument was added.

Usage Guidelines

The user can control the contents of the standardized header. The display options for the header are as follows:

- Short 6-byte GUID
- Full 16-byte GUID
- Short header which contains only the CallEntry ID

The format of the GUID headers are as follows: **//CallEntryID/GUID/Module-Dependent-List/Function-name:**.

The format of the short header is as follows: **//CallEntryID/Function-name:**.

When the **voice call debug short-header** command is entered, the header displays with no GUID or module-specific parameters. When the **no voice call debug short-header** command is entered, the header, the 6-byte GUID, and module-dependent parameter output displays. The default option is displaying the 6-byte GUID trace.

Note Using the **no** form of this command does not turn off the debugging.

Examples

The following is sample output when the **full-guid** keyword is specified:

```
Router# voice call debug full-guid
!
00:05:12: //1/0E2C8A90-BC00-11D5-8002-DACCFDCEF87D/VTSP:(0:D):0:0:4385/vtsp_insert_cdb:
00:05:12: //-1/xxxxxxxx-xxxx-xxxx-xxxx-xxxxxxxxxxxx/CCAPI/cc_incr_if_call_volume:
00:05:12: //1/0E2C8A90-BC00-11D5-8002-DACCFDCEF87D/VTSP:(0:D):0:0:4385/vtsp_open_voice_and
_set_params:
00:05:12: //1/0E2C8A90-BC00-11D5-8002-DACCFDCEF87D/VTSP:(0:D):0:0:4385/vtsp_modem_proto_fr
om_cdb:
00:05:12: //1/0E2C8A90-BC00-11D5-8002-DACCFDCEF87D/VTSP:(0:D):0:0:4385/set_playout_cdb:
00:05:12: //1/0E2C8A90-BC00-11D5-8002-DACCFDCEF87D/VTSP:(0:D):0:0:4385/vtsp_dsp_echo_cance
ller_control:
```

Note The "//-1/" output indicates that CallEntryID for the CCAPI module is not available.

Table 161 describes the significant fields shown in the display.

Table 161 voice call debug full-guid Field Descriptions

Field	Description
VTSP:(0:D):0:0:4385	Identifies the VTSP module, port name, channel number, DSP slot, and DSP channel number.
vtsp_insert_cdb	Identifies the function name.
CCAPI	Identifies the CCAPI module.

The following is sample output for the **voice call debug** command when the **short-header** keyword is specified:

```
Router(config)# voice call debug short-header
!
00:05:12: //1/vtsp_insert_cdb:
00:05:12: //-1/cc_incr_if_call_volume:
00:05:12: //1/vtsp_open_voice_and_set_params:
00:05:12: //1/vtsp_modem_proto_from_cdb:
00:05:12: //1/set_playout_cdb:
00:05:12: //1/vtsp_dsp_echo_canceller_control:
```

Note The output "//-1/" indicates that CallEntryID for CCAPI is not available.

Related Commands

Command	Description
debug rtsp api	Displays debug output for the RTSP client API.
debug rtsp client session	Displays debug output for the RTSP client data.
debug rtsp error	Displays error message for RTSP data.
debug rtsp pmh	Displays debug messages for the PMH.
debug rtsp socket	Displays debug output for the RTSP client socket data.
debug voip ccapi error	Traces error logs in the CCAPI.
debug voip ccapi inout	Traces the execution path through the CCAPI.
debug voip ivr all	Displays all IVR messages.
debug voip ivr applib	Displays IVR API libraries being processed.
debug voip ivr callsetup	Displays IVR call setup being processed.
debug voip ivr digitcollect	Displays IVR digits collected during the call.
debug voip ivr dynamic	Displays IVR dynamic prompt play debug.
debug voip ivr error	Displays IVR errors.
debug voip ivr script	Displays IVR script debug.
debug voip ivr settlement	Displays IVR settlement activities.
debug voip ivr states	Displays IVR states.
debug voip ivr tclcommands	Displays the TCL commands used in the script.
debug voip rawmsg	Displays the raw VoIP message.
debug vtsp all	Enables **debug vtsp session**, **debug vtsp error**, and **debug vtsp dsp**.
debug vtsp dsp	Displays messages from the DSP.
debug vtsp error	Displays processing errors in the VTSP.
debug vtsp event	Displays the state of the gateway and the call events.
debug vtsp port	Limits VTSP debug output to a specific voice port.
debug vtsp rtp	Displays the voice telephony RTP packet debugging.
debug vtsp send-nse	Triggers the VTSP software module to send a triple redundant NSE.
debug vtsp session	Traces how the router interacts with the DSP.
debug vtsp stats	Debugs periodic statistical information sent and received from the DSP
debug vtsp vofr subframe	Displays the first 10 bytes of selected VoFR subframes for the interface.
debug vtsp tone	Displays the types of tones generated by the VoIP gateway.

voice call send-alert

To enable the terminating gateway to send an alert message instead of a progress message after it receives a call setup message, use the **voice call send-alert command in** global configuration mode. To reset to the default, use the **no** form of this command.

voice call send-alert

no voice call send-alert

Syntax Description

This command has no arguments or keywords.

Defaults

The terminating gateway sends a progress message after it receives a call Setup message.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)XI4	This command was introduced.
12.1(5)T	This command was not supported in this release.
12.1(5.3)T	This command was integrated into this release.
12.2(1)	This command was integrated into this release.

Usage Guidelines

In Cisco IOS Release 12.1(3)XI and later, the terminating gateway sends a Progress message with a progress indicator (PI) after it receives a Setup message. Previously, the gateway responded with an Alert message after receiving a call. In some cases, if the terminating switch does not forward the progress message to the originating gateway, the originating gateway does not cut-through the voice path until a Connect is received and the caller does not hear a ringback tone. In these cases, you can use the **voice call send-alert** command to make the gateway backward compatible with releases earlier than Cisco IOS Release 12.1(3)XI. If you configure the **voice call send-alert** command, the terminating gateway sends an Alert message after it receives a Setup message from the originating gateway.

To complete calls from a PRI to an FXS interface, configure the **voice call send-alert** command on the FXS device.

Examples

The following example configures the gateway to send an Alert message:

```
voice call send-alert
```

Related Commands

Command	Description
progress_ind	Sets a specific PI in call Setup, Progress, or Connect messages from an H.323 VoIP gateway.

voice call trigger hwm

To set the high water mark in the available capacity in the trunk group or prefix databases, use the **voice call trigger hwm** command in global configuration mode. To disable the trigger point, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **trigger hwm** *percent*

no voice call {**carrier** | **trunk-group** | **prefix**} **trigger hwm** *percent*

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
percent	High-watermark value, as a percentage. Range is from 50 to 100. Default is 80. If set to 100, this trigger turns off.

Defaults

80 percent

Command Modes

Global configuration.

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

Available circuits are reported when the value of AC goes above a threshold, called the *high water mark*. This can be configured with the **voice call trigger hwm** command. When the **hwm** option is selected and the value is set to 100, no update is sent due to high water mark.

Examples

The following example sets the trigger for available capacity on trunk groups to send at a high water mark of 75%:

```
Router(config)# voice call trunk-group trigger hwm 75
```

Related Commands

Command	Description
voice call capacity mir	Sets the values for the minimum interval between reporting (MIR) and smoothing transition time for weight (STW).
voice call capacity reporting	Turns on the reporting of maxima (first derivative) or inflection (second derivative) points in available capacity.
voice call capacity timer interval	Sets the periodic interval for reporting capacity from carrier, trunk group, or prefix databases

Command	Description
voice call trigger lwm	Sets the value for low water mark in the available capacity for carrier, trunk group, or prefix databases
voice call trigger percent-change	Sets the value for percentage change in the available capacity for carrier, trunk group, or prefix databases

voice call trigger lwm

To set the value for low water mark in the available capacity in the trunk group or prefix databases, use the **voice call trigger lwm** command in global configuration mode. To disable the trigger point, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **trigger lwm** *percent*

no voice call {**carrier** | **trunk-group** | **prefix**} **trigger lwm** *percent*

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
percent	Low-watermark value, as a percentage. Range is from 0 to 30. Default is 10. If set to 0, this trigger turns off.

Defaults

10 percent

Command Modes

Global configuration.

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

Available circuits are reported when the value of AC falls below a threshold, called the *low water mark*. When the **lwm** option is selected and the value is set to 0, no update is sent due to low water mark.

Examples

The following example sets the trigger for available capacity for E.164 prefixes to send at a low water mark of 25%:

```
Router(config)# voice call prefix trigger lwm 25
```

Related Commands

Command	Description
voice call capacity mir	Sets the values for the minimum interval between reporting (MIR) and smoothing transition time for weight (STW).
voice call capacity reporting	Turns on the reporting of maxima (first derivative) or inflection (second derivative) points in available capacity.
voice call capacity timer interval	Sets the periodic interval for reporting capacity from carrier, trunk group, or prefix databases

Command	Description
voice call trigger hwm	Sets the value for high water mark in the available capacity for carrier, trunk group, or prefix databases
voice call trigger percent-change	Sets the value for percentage change in the available capacity for carrier, trunk group, or prefix databases

voice call trigger percent-change

To set the percentage change in the available capacity in the trunk group or prefix databases, use the **voice call trigger** command in global configuration mode. To disable the trigger point, use the **no** form of this command.

voice call {**carrier** | **trunk-group** | **prefix**} **trigger percent-change** *percent*

no voice call {**carrier** | **trunk-group** | **prefix**} **trigger percent-change** *percent*

Syntax Description

carrier	Carrier-code address family.
trunk-group	Trunk-group address family.
prefix	E.164 prefix.
percent	Percentage change. Range is from 0 to 100. Default is 30. If set to 0, this trigger turns off.

Defaults

30 percent

Command Modes

Global configuration.

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

Available circuits are reported when the absolute percent change is above a threshold. When the **percent-change** option is selected and the value is set to 0, no update for percent change is sent

Examples

The following example sets the trigger for available capacity on the carrier codes to send at a percentage change of 15%:

```
Router(config)# voice call carrier trigger percent-change 15
```

Related Commands

Command	Description
voice call capacity mir	Sets the values for the minimum interval between reporting (MIR) and smoothing transition time for weight (STW).
voice call capacity reporting	Turns on the reporting of maxima (first derivative) or inflection (second derivative) points in available capacity.
voice call capacity timer interval	Sets the periodic interval for reporting capacity from carrier, trunk group, or prefix databases

Command	Description
voice call trigger hwm	Sets the value for high water mark in the available capacity for carrier, trunk group, or prefix databases
voice call trigger lwm	Sets the value for low water mark in the available capacity for carrier, trunk group, or prefix databases

voice class aaa

To enable dial-peer-based VoIP AAA configurations, use the **voice class aaa** command in global configuration mode. To disable dial-peer-based VoIP AAA configurations, use the **no** form of this command.

voice class aaa *tag*

no voice class aaa *tag*

Syntax Description

tag	Voice-class AAA identifier. Range is from 1 to 10000. There is no default.

Defaults

No default behaviors or values

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

The **voice class aaa** configuration command sets up a voice service class that allows you to perform dial-peer-based AAA configurations.

The command activates voice class AAA configuration mode. Commands that are configured in voice class AAA configuration mode are listed in the "Related Commands" section.

Examples

The following example shows AAA configurations in voice class AAA configuration mode. The number assigned to the tag is 1.

```
voice class aaa 1
 authentication method dp
 authorization method dp
 accounting method dp
in-bound
 accounting template temp-dp
```

The following example shows accounting configurations in voice class AAA configuration mode:

```
voice class aaa 2
 accounting method dp-out out-bound
 accounting template temp-dp out-bound
```

Related Commands

Command	Description
authentication method	Specifies an authentication method for calls coming into the defined dial peer.
authorization method	Specifies an authorization method for calls coming into the defined dial peer.
method	Specifies an accounting method for calls coming into the defined dial peer.
accounting suppress	Disables accounting that is automatically generated by the service provider module for a specific dial peer.
voice-class aaa	Applies properties defined in the voice class to a specific dial peer.

voice-class aaa (dial peer)

To apply properties defined in the voice class to a dial peer, use the **voice-class aaa** command in dial peer configuration mode. This command does not have a **no** form.

voice-class aaa *tag*

Syntax Description

tag	Voice-class AAA identifier. Range is from 1 to 10000. There is no default.

Defaults

No default behaviors or values

Command Modes

Dial peer configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

Properties that are configured in voice class AAA configuration mode can be applied to a dial peer by using the **voice-class aaa** command in dial peer configuration mode.

Examples

The following example shows redirecting AAA requests using Digital Number Identification Service (DNIS). You define a voice class to specify the AAA methods and then use the **voice-class aaa** command in dial peer configuration mode.

```
voice class aaa 1
  authentication method kz
  authorization method kz
  accounting method kz
!
dial-peer voice 100 voip
  incoming called-number 50..
  session target ipv4:1.5.31.201
  voice-class aaa 1
```

Related Commands

Command	Description
voice class aaa	Enables dial-peer-based VoIP AAA configurations.

voice class busyout

To create a voice class for local voice busyout functions, use the **voice class busyout command in** global configuration mode. To delete the voice class, use the **no** form of this command.

voice class busyout *tag*

no voice class busyout *tag*

Syntax Description

tag	Unique identifier assigned to one voice class. Range is from 1 to 10000. There is no default.

Defaults

No voice class is configured for busyout functions.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600, Cisco 3600, and Cisco MC3810.

Usage Guidelines

You can apply a busyout voice class to multiple voice ports. You can assign only one busyout voice class to a voice port. If a second busyout voice class is assigned to a voice port, the second voice class replaces the one previously assigned.

If you assign a busyout voice class to a voice port, you may not assign separate busyout commands directly to the voice port, such as **busyout monitor serial**, **busyout monitor ethernet**, or **busyout monitor probe**.

Examples

The following example configures busyout voice class 20, in which the connections to two remote interfaces are monitored by a response time reporter (RTR) probe with a G.711ulaw profile, and voice ports are busied out whenever both links have a packet loss exceeding 10 percent and a packet delay time exceeding 2 seconds:

```
voice class busyout 20
 busyout monitor probe 171.165.202.128 g711u loss 10 delay 2000
 busyout monitor probe 171.165.202.129 g711u loss 10 delay 2000
```

The following example configures busyout voice class 30, in which voice ports are busied out when serial ports 0/0, 1/0, 2/0, and 3/0 go out of service.

```
voice class busyout 30
 busyout monitor serial 0/0
 busyout monitor serial 1/0
 busyout monitor serial 2/0
 busyout monitor serial 3/0
```

Related Commands

Command	Description
busyout monitor ethernet	Configures a voice port to monitor a local Ethernet interface for events that would trigger a voice-port busyout.
busyout monitor probe	Configures a voice port to enter the busyout state if an RTR probe signal returned from a remote, IP-addressable interface crosses a specified delay or loss threshold.
busyout monitor serial	Configures a voice port to monitor a serial interface for events that would trigger a voice-port busyout.
show voice busyout	Displays information about the voice busyout state.

voice class codec

To enter voice-class configuration mode and assign an identification tag number for a codec voice class, use the **voice class codec** command in global configuration mode. To delete a codec voice class, use the **no** form of this command.

voice class codec *tag*

no voice class codec *tag*

Syntax Description

tag	Unique identifier assigned to the voice class. Range is from 1 to 10000. There is no default.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced on the Cisco AS5300.
12.0(7)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.0(7)XK	This command was implemented on the Cisco MC3810.
12.1(2)T	This command was integrated into this release.

Usage Guidelines

This command only creates the voice class for codec selection preference and assigns an identification tag. Use the **codec preference** command to specify the parameters of the voice class, and use the **voice-class codec** dial-peer command to apply the voice class to a Voice over IP (VoIP) dial peer.

Note The **voice class codec** command in global configuration mode is entered without the hyphen. The **voice-class codec** command in dial-peer configuration mode is entered with the hyphen.

Examples

The following example shows how to enter voice-class configuration mode and assign a voice class tag number starting from global configuration mode:

```
voice class codec 10
```

After you enter voice-class configuration mode for codecs, use the **codec preference** command to specify the parameters of the voice class.

The following example creates preference list 99, which can be applied to any dial peer:

```
voice class codec 99
 codec preference 1 g711alaw
 codec preference 2 g711ulaw bytes 80
```

```
codec preference 3 g723ar53
codec preference 4 g723ar63 bytes 144
codec preference 5 g723r53
codec preference 6 g723r63 bytes 120
codec preference 7 g726r16
codec preference 8 g726r24
codec preference 9 g726r32 bytes 80
codec preference 10 g728
codec preference 11 g729br8
codec preference 12 g729r8 bytes 50
```

Related Commands

Command	Description
codec preference	Specifies a list of preferred codecs to use on a dial peer.
test voice port detector	Defines the order of preference in which network dial peers select codecs.
voice-class codec (dial peer)	Assigns a previously configured codec selection preference list to a dial peer.

voice-class codec (dial peer)

To assign a previously configured codec selection preference list (codec voice class) to a Voice over IP (VoIP) dial peer, enter the **voice-class codec command in** dial-peer configuration mode. To remove the codec preference assignment from the dial peer, use the **no** form of this command.

voice-class codec *tag*

no voice-class codec *tag*

Syntax Description

tag	Unique identifier assigned to the voice class. Range is from 1 to 10000. The *tag* number maps to the tag number created using the **voice class codec** global configuration command.

Defaults

Dial peers have no codec voice class assigned.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced on the Cisco AS5300.
12.0(7)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.0(7)XK	This command was implemented on the Cisco MC3810.
12.1(2)T	This command was integrated into this release.

Usage Guidelines

You can assign one voice class to each VoIP dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

Note The **voice-class codec** command in dial-peer configuration mode is entered with a hyphen. The **voice class codec** command in global configuration mode is entered without a hyphen.

Examples

The following example shows how to assign a previously configured codec voice class to a dial peer:

```
dial-peer voice 100 voip
 voice-class codec 10
```

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all dial peers configured on the router.

test voice port detector	Defines the order of preference in which network dial peers select codecs.
voice class codec	Enters voice-class configuration mode and assigns an identification tag number for a codec voice class.

voice class custom-cptone

To create a voice class for defining custom call-progress tones to be detected, use the **voice class custom-cptone** command in global configuration mode. To delete the voice class, use the **no** form of this command.

voice class custom-cptone *cptone-name*

no voice class custom-cptone *cptone-name*

Syntax Description

cptone-name	Descriptive identifier for this class of custom call-progress tones that associates this set of custom call-progress tones with voice ports.

Defaults

No voice class of custom call-progress tones is created.

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)XM	This command was introduced on the Cisco 2600, Cisco 3600, and Cisco MC3810.
12.2(2)T	This command was implemented on the Cisco 1750.

Usage Guidelines

After you create a voice class, you need to define custom call-progress tones for this voice class using the **dualtone** command.

Examples

The following example creates a voice class named country-x.

```
voice class custom-cptone country-x
```

The following example deletes the voice class named country-x.

```
no voice class custom-cptone country-x
```

Related Commands

Command	Description
dualtone	Defines the tone and cadence for a custom call-progress tone.
supervisory custom-cptone	Associates a class of custom call-progress tones with a voice port.
voice class dualtone-detect-params	Modifies the boundaries and limits for call-progress tones.

voice class dualtone

To create a voice class for Foreign Exchange Office (FXO) supervisory disconnect tone detection parameters, use the **voice class dualtone** command in global configuration mode. To delete the voice class, use the **no** form of this command.

voice class dualtone *tag*

no voice class dualtone *tag*

Syntax Description

tag	Unique identifier assigned to one voice class. Range is from 1 to 10000. There is no default.

Defaults

No voice class is configured for tone detection parameters.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600 series, Cisco 3600, and Cisco MC3810.

Usage Guidelines

Use this command first to create the voice class. Then use the **supervisory disconnect dualtone voice-class** command to assign the voice class to a voice port.

A voice class can define any number of tones to be detected. You need to define a matching tone for each supervisory disconnect tone expected from a PBX or from the public switched telephone network (PSTN).

Examples

The following example configures voice class dualtone 70, which defines one tone with two frequency components, and does not configure a cadence list:

```
voice class dualtone 100
 freq-pair 1 350 440
 freq-max-deviation 10
 freq-max-power 6
 freq-min-power 25
 freq-power-twist 15
 freq-max-delay 16
 cadence-min-on-time 50
 cadence-max-off-time 400
 cadence-variation 8
 exit
```

The following example configures voice class dualtone 100, which defines one tone with two frequency components, and configures a cadence list:

```
voice class dualtone 100
```

```
 freq-pair 1 350 440
 freq-pair 2 480 850
 freq-max-deviation 10
 freq-max-power 6
 freq-min-power 25
 freq-power-twist 15
 freq-max-delay 16
 cadence-min-on-time 50
 cadence-max-off-time 400
 cadence-list 1 100 100 300 300
 cadence-variation 8
 exit
```

The following example configures voice class dualtone 90, which defines three tones, each with two frequency components, and configures two cadence lists:

```
voice class dualtone 90
 freq-pair 1 350 440
 freq-pair 2 480 850
 freq-pair 3 1000 1250
 freq-max-deviation 10
 freq-max-power 6
 freq-min-power 25
 freq-power-twist 15
 freq-max-delay 16
 cadence-min-on-time 50
 cadence-max-off-time 500
 cadence-list 1 100 100 300 300 100 200
 cadence-list 2 100 200 100 400
 cadence-variation 8
 exit
```

Related Commands

Command	Description
supervisory disconnect dualtone voice-class	Assigns a previously configured voice class for FXO supervisory disconnect tone to a voice port.

voice class dualtone-detect-params

To create a voice class for defining a set of tolerance limits for the frequency, power, and cadence parameters of the tones to be detected, use the **voice class dualtone-detect-params** command in global configuration mode. To delete the voice class, use the **no** form of this command.

voice class dualtone-detect-params *tag*

no voice class dualtone-detect-params *tag*

Syntax Description

tag	Unique identifier assigned to a voice class. Range is from 1 to 10000. There is no default.

Defaults

No voice class is configured for defining answer-supervision tolerance limits.

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)XM	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)T	This command was implemented on the Cisco 1750.

Usage Guidelines

Use this command to create a voice class in which you can define maximum and minimum call-progress tone tolerance parameters that you can apply to any voice port. These parameters further define the call-progress tones defined by the **voice class custom-cptone** command. Use the **supervisory dualtone-detect-params** command to apply these tolerance parameters to a voice port.

Examples

The following example creates voice class 70, in which you can specify modified boundaries and limits for call-progress tone detection.

```
voice class dualtone-detect-params 70
freq-max-deviation 25
freq-max-power -5
freq-min-power -20
freq-power-twist 10
freq-max-delay 50
cadence-variation 80
exit
```

Related Commands

Command	Description
supervisory dualtone-detect-params	Assigns the boundary and detection tolerance parameters defined by the **voice class dualtone-detect-params** command to a voice port.
voice class custom-cptone	Creates a voice class for defining custom call-progress tones.

voice class h323

To create an H.323 voice class that is independent of a dial peer and can be used on multiple dial peers, use the **voice class h323** command in global configuration mode. To remove the voice class, use the **no** form of this command.

voice class h323 *tag*

no voice class h323

Syntax Description

tag	Unique identifier assigned to the voice class. Range is from 1 to 10000. There is no default.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on the Cisco 1700, Cisco 2600 series, Cisco 3600 series, Cisco 7200, Cisco AS5300, Cisco uBR910, and Cisco uBR924.

Usage Guidelines

The **voice class h323** command in global configuration mode does not include a hyphen. The **voice-class h323** command in dial-peer configuration mode includes a hyphen.

Examples

The following example creates an H.323 voice class labeled 1:

```
voice class h323 1
```

Related Commands

Command	Description
h225 timeout tcp establish	Sets the H.225 TCP timeout value.

voice-class h323 (dial peer)

To assign an H.323 voice class to a VoIP dial peer, use the **voice-class h323** command in dial-peer configuration mode. To remove the voice class from the dial peer, use the **no** form of this command.

voice-class h323 *tag*

no voice-class h323 *tag*

Syntax Description

tag	Unique identifier assigned to the voice class. Range is from 1 to 10000. There is no default.

Defaults

The dial peer does not use an H.323 voice class.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(2)T	This command was introduced.

Usage Guidelines

The voice class that you assign to the dial peer must be configured using the **voice class h323** in global configuration mode.

You can assign one voice class to each VoIP dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

The **voice-class h323** command in dial-peer configuration mode includes a hyphen and in global configuration mode does not include a hyphen.

Examples

The following example shows how to create an H.323 voice class and then assign it to a dial peer:

```
voice class h323 10
dial-peer voice 100 voip
 voice-class h323 10
```

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all dial peers configured on the router.
voice class h323	Enters voice-class configuration mode and assigns an identification tag number for an H.323 voice class.

voice class permanent

To create a voice class for a Cisco trunk or FRF.11 trunk, use the **voice class permanent** command in global configuration mode. To delete the voice class, use the **no** form of this command.

voice class permanent *tag*

no voice class permanent *tag*

Syntax Description

tag	Unique identifier assigned to the voice class. Range is from 1 to 10000. There is no default.

Defaults

No voice class is configured.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810.
12.0(4)T	This command was integrated into this release.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.

Usage Guidelines

The **voice class permanent** command can be used for Voice over Frame Relay (VoFR), Voice over ATM (VoATM), and Voice over IP (VoIP) trunks.

The **voice class permanent** command in global configuration mode is entered without a hyphen. The **voice-class permanent** command in dial-peer and voice-port configuration modes is entered with a hyphen.

Examples

The following example shows how to create a permanent voice class starting from global configuration mode:

```
voice class permanent 10
 signal keepalive 3
 exit
```

Related Commands

Command	Description
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal timing oos	Configures the signal timing parameter for the OOS state of a call.
signal-type	Sets the signaling type for a network dial peer.
voice-class permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a network dial peer.

voice-class permanent (dial-peer)

To assign a previously configured voice class for a Cisco trunk or FRF.11 trunk to a network dial peer, use the **voice-class permanent** command in dial-peer configuration mode. To remove the voice-class assignment from the network dial peer, use the **no** form of this command.

voice-class permanent *tag*

no voice-class permanent *tag*

Syntax Description

tag	Unique identifier assigned to the voice class. The *tag* number maps to the tag number created using the **voice class permanent** global configuration command. Range is from 1 to 10000. There is no default.

Defaults

Network dial peers have no voice class assigned.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810.
12.0(4)T	This command was integrated into this release.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.

Usage Guidelines

You can assign one voice class to any given network dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

You cannot assign a voice class to a plain old telephone service (POTS) dial peer.

The **voice-class permanent** command in dial-peer configuration mode is entered with a hyphen. The **voice class permanent** command in global configuration mode is entered without a hyphen.

Examples

The following example assigns a previously configured voice class to a Voice over Frame Relay (VoFR) network dial peer:

```
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal timing oos	Configures the signal timing parameter for the OOS state of a call.
signal-type	Sets the signaling type for a network dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.

voice-class permanent (voice-port)

To assign a previously configured voice class for a Cisco trunk or FRF.11 trunk to a voice port, use the **voice-class permanent** command in voice-port configuration mode. To remove the voice-class assignment from the voice port, use the **no** form of this command.

voice-class permanent *tag*

no voice-class permanent *tag*

Syntax Description

tag	Unique identifier assigned to the voice class. The *tag* number maps to the tag number created using the **voice class permanent** global configuration command. Range is 1 to 10000. There is no default.

Defaults

Voice ports have no voice class assigned.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810.
12.0(4)T	This command was integrated into this release.
12.1(3)T	This command was implemented as a voice-port configuration command on the Cisco 2600 series and Cisco 3600 series.

Usage Guidelines

You can assign one voice class to any given voice port. If you assign another voice class to a voice port, the last voice class assigned replaces the previous voice class.

The **voice-class permanent** command in voice-port configuration mode is entered with a hyphen. The **voice class permanent** command in global configuration mode is entered without a hyphen.

Examples

The following example assigns a previously configured voice class to voice port 1/1 in a Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
 voice-class permanent 10
```

The following example assigns a previously configured voice class to voice port 1/1/0 in a Cisco 3600 series router:

```
voice-port 1/1/0
 voice-class permanent 10
```

Related Commands

Command	Description
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal timing oos	Configures the signal timing parameter for the OOS state of a call.
signal-type	Sets the signaling type for a network dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.

voice confirmation-tone

To disable the two-beep confirmation tone for private line, automatic ringdown (PLAR), or PLAR off-premises extension (OPX) connections, use the **voice confirmation-tone** command in voice-port configuration mode. To enable the two-beep confirmation tone, use the **no** form of this command.

voice confirmation-tone

no voice confirmation-tone

Syntax Description

This command has no arguments or keywords.

Defaults

The two-beep confirmation tone is heard on PLAR and PLAR OPX connections.

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

This command applies only to the Cisco MC3810 multiservice concentrator.

Use this command to disable the two-beep confirmation tone that a caller hears when picking up the handset for PLAR and PLAR OPX connections. This command is valid only if the voice-port **connection** command is set to PLAR or PLAR OPX.

Examples

The following example disables the two-beep confirmation tone on voice port 1/1 on the Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
 connection plar-opx
 voice confirmation-tone
```

Related Commands

Command	Description
connection	Specifies a connection mode for a voice port.

voice dnis-map

To create or modify a Digital Number Identification Service (DNIS) map, use the **voice dnis-map** command in global configuration mode. To delete a DNIS map, use the **no** form of this command.

voice dnis-map *map-name* [*url*]

no voice dnis-map *map-name*

Syntax Description

map-name	Name of the DNIS map.
url	(Optional) URL of an external text file that contains a list of DNIS entries.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

A DNIS map is a table of DNIS numbers associated with a single dial peer. For applications such as VoiceXML, using a DNIS map makes it possible to configure a single dial peer for all DNIS numbers used to refer to VoiceXML documents. Keep the following considerations in mind when using voice DNIS maps.

- A separate entry must be made for each DNIS entry in a DNIS map. Wildcards are not supported.
- If a URL is not supplied, the command enters DNIS-map configuration mode, permitting the entry of DNIS numbers by using the **dnis** command.
- The URL argument points to the location of an external text file containing a list of DNIS entries (for example: tftp://dnismap.txt). This allows the administrator to maintain a single master file of all DNIS map entries, if desired, rather than configuring the DNIS entries on each gateway.

  The name of the text file extension is not significant; .doc, .txt, or .cfg are all acceptable because the extension is not checked. The entries in the file should look the same as a DNIS entry configured in Cisco IOS software (for example: **dnis 5553305 url tftp://global/tickets/movies.vxml**).
- External text files used for DNIS maps must be stored on TFTP servers; they cannot be stored on HTTP servers.
- To associate a DNIS map with a dial peer, use the **dnis-map** command.
- To view the configuration information for DNIS maps, use the **show voice dnis-map** command.

Examples

The following example shows how the **voice dnis-map** command is used to create a DNIS map:

```
voice dnis-map dmap1
```

The following example shows the **voice dnis-map** command used with a URL that specifies the location of a text file containing the DNIS entries:

```
voice dnis-map dmap2 tftp://keyer/dmap2/dmap2.txt
```

Following is an example of the contents of a text file comprising a DNIS map:

```
!Example dnis-map with 8 entries.
!
dnis 5551212 url tftp://global/ticket/vapptest1.vxml
dnis 5551111 url tftp://global/ticket/vapptest2.vxml
dnis 5551234 url tftp://global/ticket/vapptest3.vxml
dnis 5556789
dnis 5552000
dnis 5552100
dnis 5552200
dnis 5552300
```

Related Commands

Command	Description
dnis	Adds a DNIS number to a DNIS map.
dnis-map	Associates a DNIS map with a dial peer.
show voice dnis-map	Displays configuration information about DNIS maps.
voice dnis-map load	Reloads a DNIS map that has changed since the previous load.

voice dnis-map load

To reload a DNIS map that has been modified, use the **voice dnis-map load** command in privileged EXEC mode.

voice dnis-map load *map-name*

Syntax Description

map-name	Name of the DNIS map to reload.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command reloads a DNIS map residing on an external server. Use this command when the DNIS map file has changed since the previous load.

To create or modify a DNIS map, use the **voice dnis-map** command.

Examples

The following example shows how the **voice dnis-map load** command is used to reload a DNIS map named “mapfile1”:

```
Router# voice dnis-map load mapfile1
```

Related Commands

Command	Description
dnis	Adds a DNIS number to a DNIS map.
dnis-map	Associates a DNIS map with a dial peer.
show voice dnis-map	Displays configuration information about DNIS maps.
voice dnis-map	Enters DNIS map configuration mode to create a DNIS map.

voice echo-canceller extended

To enable the G.168 extended echo canceller (EC) on the Cisco 1700 series or Cisco ICS7750, use the **voice echo-canceller extended** command in global configuration mode. To return to the Cisco-proprietary G.165 default EC, use the **no** form of this command.

voice echo-canceller extended

no voice echo-canceller extended

Syntax Description

This command has no arguments or keywords.

Defaults

The G.168 extended EC is not enabled.

Command Modes

Global configuration

Command History

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

Usage Guidelines

You do not have to shut down all the voice ports on the Cisco 1700 series or Cisco ICS7750 in order to switch the echo canceller, but you should make sure that when you switch the echo canceller, there are no active calls on the router.

Because echo cancellation is an invasive process that can minimally degrade voice quality, this command should be disabled if it is not needed.

Note This command is valid only when the **echo-canceller coverage** command has been configured.

Examples

To switch to the G.168 extended EC from the Cisco default EC on the Cisco 1700 series or Cisco ICS7750 platforms, use the following command in global configuration mode:

```
Router(config)# voice echo-canceller extended
```

Related Commands

Command	Description
echo-cancel enable	Enables the cancellation of voice that is sent and received on the same interface.
echo-canceller coverage	Adjusts the size of the EC and selects the extended EC when the Cisco default EC is present.

voice enum-match-table

To create an ENUM match table for voice calls, use the **voice enum-match-table** in global configuration mode. To delete the ENUM match table, use the **no** form of this command.

voice enum-match-table *table-number*

no voice enum-match-table *table-number*

Syntax Description

table-number	Number of the ENUM match table. Range is from 1 to 15. There is no default.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

The ENUM match table is a set of rules for matching incoming calls. When a call comes in, its called number is matched against the match pattern of the rule with the highest preference.

If it matches, the replacement pattern is applied to the number. The resulting number and the domain name of the rule are used to make an ENUM query.

If the called number does not match the match pattern, the next rule in order of preference is selected.

Examples

The following example creates ENUM match table 3 for voice calls:

```
Router(config)# voice enum-match-table 3
Router(config-enum)# rule 1 5/(.*)/ /\1/e164.cisco.com
Router(config-enum)# rule 2 4/^9011\(.*\)/ /\1/e164.arpa
```

In this table, rule 1 matches any number. The resulting number is the same as the called number. That number and the domain name “e164.cisco.com” are used to make an ENUM query.

Rule 2 matches any number that starts with 9011. The 9011 is removed from the incoming number. The resulting number and the domain name “e164.arpa” are used for the ENUM query.

Suppose an incoming call has a called number of 4085551212. [Rule 2 is applied] first because it has a higher preference. The first few digits, 4085, do not match the 9011 pattern of rule 2, so [rule 1 is applied] next. The called number matches rule 1, and the resulting number is 4085551212. This number and “e164.cisco.com” form the ENUM query (2.1.2.1.5.5.5.8.0.4.e164.cisco.com).

Related Commands

Command	Description
rule (ENUM configuration)	Defines the matching, replacement, and rejection patterns for an ENUM match table.
show voice enum-match-table	Displays the configuration of voice ENUM match tables.
test enum	Tests the functionality of an ENUM match table.

voice hpi capture

To allocate the Host Port Interface (HPI) capture buffer size (in bytes) and to set up or change the destination URL for captured data, use the **voice hpi capture** command in global configuration mode. To stop all logging and file operations, to disable data transport from the capture buffer, and to automatically set the buffer size to 0, use the **no** form of this command.

voice hpi capture [**buffer** *size* | **destination** *url*]

no voice hpi capture buffer *size*

Syntax Description

buffer *size*	(Optional) Size of the HPI capture buffer, in bytes. Range is from 328 to 9000000. Default is 328.
destination *url*	(Optional) Destination URL for storing captured data.

Defaults

328 bytes (no buffer is used if it is not configured explicitly)

Command Modes

Global configuration

Command History

Release	Modification
12.2(10)	This command was introduced.
12.2(11)T	This command was integrated into this release.

Usage Guidelines

If you want to change the size of an existing non-zero buffer, you must first reset it to 0 and then change it from 0 to the new size.

The **destination** *url* option sets up or changes the destination URL for captured data. To disable data transport from the capture buffer, use the **no** form of the command. If the buffer is allocated, captured data is sent to the current URL (if it was already configured) until the new URL is specified.

If a new URL differs from the current URL and logging is enabled, the current URL is closed and all further data is sent to the new URL. Entering a blank URL or prefixing the command with **no** disables data transport from the capture buffer, and (if capture is enabled) captured data is stored in the capture buffer until it reaches its capacity.

Once the buffer-queueing program is running, the transport process attempts to connect to a new or existing "capture destination" URL. A version message is written to the URL, and if the message is successfully received, any further messages placed into the message queue are written to that URL. If a new URL is entered using the **voice hpi capture destination** *url* command, the open URL is closed, and the system attempts to write to the new URL. If the new URL does not work, the transport process exits. The transport process is restarted when another URL is entered or the system is restarted.

The **buffer** *size* option sets the maximum amount of memory (in bytes) that the capture system allocates for its buffers when it is active. The capture buffer is where the captured messages are stored before they are sent to the URL specified by the capture destination. The system is started by choosing the amount of memory (greater than 0 bytes) that the buffer-queueing system can allocate to the free message pool.

HPI messages can then be captured until buffer capacity is reached. Entering **0** for the buffer size and prefixing the command with **no** stops all logging and file operations and automatically sets the buffer size to 0.

The **voice hpi capture** command can be saved with the router configuration so that the command is active during router startup. This allows you to capture the HPI messages sent during router bootup before the CLI is enabled. After you have configured the buffer size in the running configuration (valid range is from 328 to 9000000), save it to the startup configuration using the **write** command or to the TFTP server using the **copy run tftp** command.

Caution Using the message logger feature in a production network environment impacts CPU and memory usage on the gateway.

Examples

The following example changes the size (in bytes) of the HPI capture buffer and initializes the buffer-queueing program:

```
Router# configure terminal

Enter configuration commands, one per line.  End with CNTL/Z.

Router(config)# voice hpi capture buffer 40000

Router(config)# end
Router#

03:23:31:caplog:caplog_cli_interface:hpi capture buffer size set to 40000 bytes
03:23:31:caplog:caplog_logger_init:TRUE, Started task HPI Logger (PID 64)
03:23:31:caplog:caplog_cache_init:TRUE, malloc_named(39852), 123 elements (each 324 bytes
big)
03:23:31:caplog:caplog_logger_proc:Attempting to open ftp://172.23.184.233/c:b-38-117
03:23:32:%SYS-5-CONFIG_I:Configured from console by console
Router#
```

The following example sets the capture destination by entering a destination URL using FTP:

```
Router# configure terminal

Enter configuration commands, one per line.  End with CNTL/Z.

Router(config)# voice hpi capture destination ftp://172.23.184.233/c:b-38-117a
Router(config)#

04:05:10:caplog:caplog_cli_interface:hpi capture
destination:ftp://172.23.184.233/c:b-38-117a
04:05:10:caplog:caplog_logger_init:TRUE, Started task HPI Logger (PID 19)
04:05:10:caplog:caplog_cache_init:Cache must be at least 324 bytes
04:05:10:caplog:caplog_logger_proc:Terminating...

Router(config)# end
Router#
```

Related Commands

Command	Description
debug hpi	Turns on the debug output for the logger.
show voice hpi capture	Displays the capture status and statistics.

voice hunt

To configure an originating or tandem router so that it continues dial-peer hunting if it receives a user-busy disconnect code from a destination router, use the **voice hunt** command in global configuration mode. To configure the router so that it stops dial-peer hunting if it receives a user-busy disconnect code (the default option), use the **no** form of this command.

voice hunt {**user-busy** | **invalid-number** | **unassigned-number**}

no voice {**user-busy** | **invalid-number** | **unassigned-number**}

Syntax Description

user-busy	Router continues dial-peer hunting if it receives a user-busy disconnect cause code from a destination router.
invalid-number	Router stops dial-peer hunting if it receives a an invalid-number disconnect cause code from a destination router.
unassigned-number	Router stops dial-peer hunting if it receives an unassigned-number disconnect cause code from a destination router.

Defaults

The default depends on the disconnect cause code. By default, the router stops dial-peer hunting if it receives the user-busy disconnect cause code. By default, the router continues dial-peer hunting if it receives an invalid-number, or an unassigned-number disconnect cause code.

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)T	This command was introduced for VoFR on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810. It was also introduced for VoIP on Cisco 2600 series and Cisco 3600 series.
12.0(7)T	This command was implemented for VoIP on the Cisco AS5300 and Cisco AS5800.
12.0(7)XK	This command was implemented for VoIP on the Cisco MC3810.
12.1(2)T	This command was implemented for VoIP on the Cisco MC3810.
12.1(3)XI	The **invalid-number** and **unassigned-number** keywords were added, and the command name was changed to **voice hunt**.
12.1(5)T	This command was integrated into this release.

Usage Guidelines

This command applies to routers that act as originating or tandem nodes in a Voice over IP, Voice over Frame Relay, or Voice over ATM environment.

This command is used for a configuration in which an originating or tandem router is configured with multiple dial peer entries that route a call to the same destination number, but on different destination routers. In this configuration, after all routes to the first router entry in the dial-peer list are active, a new call does not "roll over" to the next router in the dial-peer list.

This failure to route to the second destination router happens when the bandwidth on the voice interface is greater than the maximum capacity of the first destination router. This condition allows the originating or tandem router to attempt to place a new call to the first destination router because it has indications from the first destination router that there is more capacity based on the bandwidth setting. When the first destination router receives the call, if all of the ports are in use, the destination router returns a “user-busy” disconnect reason code to the originating or tandem router.

The originating or tandem router interprets the disconnect reason code as “unavailable destination” for the call and returns a busy tone to the initiating caller.

The originating or tandem router fails to try other routers in the dial-peer list after receiving a “user disconnect” reason code, and so it terminates the call attempt. By using this command, you can perform dial-peer hunting on multiple destination routers even if the originating or tandem router receives a “user-busy” disconnect reason code from one of the destination routers.

Examples

The following example configures the originating or tandem router to continue dial-peer hunting if it receives a “user-busy” disconnect code from a destination router:

```
voice hunt user-busy
```

The following example configures the originating or tandem router to continue dial-peer hunting if it receives an “invalid-number” disconnect code from a destination router:

```
voice hunt invalid-number
```

Related Commands

Command	Description
huntstop	Disables all further dial-peer hunting if a call fails when using hunt groups.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.

voice local-bypass

To configure local calls to bypass the digital signal processor (DSP), use the **voice local-bypass command in** global configuration mode. To direct local calls through the DSP, use the **no** form of this command.

voice local-bypass

no voice local-bypass

Syntax Description

This command has no arguments or keywords.

Defaults

Local calls bypass the DSP.

Command Modes

Global configuration

Command History

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

Usage Guidelines

Local calls (calls between voice ports on a router or concentrator) normally bypass the DSP to minimize use of system resources. Use the **no** form of the **voice local-bypass** command if you need to direct local calls through the DSP. Input gain and output attenuation can be configured only if calls are directed through the DSP.

Examples

The following example configures a Cisco MC3810 multiservice concentrator or Cisco 2600 series or Cisco 3600 series router to pass local calls through the DSP:

```
no voice local-bypass
```

Related Commands

Command	Description
input gain	Configures a specific input gain value.
output attenuation	Configures a specific output attenuation value.

voice rtp send-recv

To establish a two-way voice path when the Real-Time Transport Protocol (RTP) channel is opened, use the **voice rtp send-recv command in** global configuration mode. To reset to the default, use the **no** form of this command.

voice rtp send-recv

no voice rtp send-recv

Syntax Description

This command has no arguments or keywords.

Defaults

The voice path is cut-through in only the backward direction when the RTP channel is opened.

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)T	This command was introduced on the Cisco 2600, Cisco 3600, Cisco 7200, Cisco 7500, Cisco AS5300, Cisco AS5800, and Cisco MC3810.
12.2(2)XA	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(11)T	This command was integrated into this release.

Usage Guidelines

This command should be enabled only when the voice path must be cut-through (established) in both the backward and forward directions before a Connect message is received from the destination switch. This command affects all VoIP calls when it is enabled.

Examples

The following example enables the voice path to cut-through in both directions when the RTP channel is opened:

```
voice rtp send-recv
```

voice service

To enter voice-service configuration mode and to specify a voice-encapsulation type, use the **voice service** command in global configuration mode.

voice service {**pots** | **voatm** | **vofr** | **voip**}

Syntax Description

pots	Telephony voice service.
voatm	Voice over ATM (VoATM) encapsulation.
vofr	Voice over Frame Relay (VoFR) encapsulation.
voip	Voice over IP (VoIP) encapsulation.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)XA	This command was introduced on the Cisco MC3810.
12.1(2)T	This command was integrated into this release.
12.1(3)T	This command was implemented for VoIP on the Cisco 2600 series and Cisco 3600 series.
12.1(3)XI	This command was implemented on the Cisco AS5300.
12.1(5)T	This command was integrated into this release.
12.1(5)XM	This command was implemented on the Cisco AS5800.
12.1(5)XM2	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(2)T	This command was implemented on the Cisco 7200 series.
12.2(11)T	This command was implemented on the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

Usage Guidelines

Voice-service configuration mode is used for packet telephony service commands that affect the gateway globally.

Examples

The following example enters voice-service configuration mode for VoATM service commands:

```
voice service voatm
```

voice source-group

To define a source IP group for voice calls, use the **voice source-group** command in global configuration mode. To delete the source IP group, use the **no** form of this command.

voice source-group *name*

no voice source-group *name*

Syntax Description

name	Name of the IP group. Maximum length is 31 alphanumeric characters.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the **voice source-group** command to assign a name to a set of source IP group characteristics. The terminating gateway uses these characteristics to identify and translate the incoming VoIP call.

Carrier IDs and trunk group labels must not have the same names.

Do not mix carrier IDs and trunk group labels within a source IP group.

A terminating gateway can be configured with carrier ID source IP groups and trunk-group-label source IP groups. The name of the source IP group must be unique to the gateway.

Examples

The following example initiates source IP group "utah2" for VoIP calls:

```
Router(config)# voice source-group utah2
```

Related Commands

Command	Description
access-list	Defines a list of source groups for identifying incoming calls.
carrier-id (voice source group)	Specifies the carrier handling a VoIP call.
description (voice source group)	Assigns a disconnect cause to a source IP group.
h323zone-id (voice source group)	Assigns a zone ID to an incoming H.323 call.

Command	Description
translation-profile (source group)	Assigns a translation profile to a source IP group.
trunk-group-label (voice source group)	Specifies the trunk handling a VoIP call.

voice translation-profile

To define a translation profile for voice calls, use the **voice translation-profile** command in global configuration mode. To delete the translation profile, use the **no** form of this command.

voice translation-profile *name*

no voice translation-profile *name*

Syntax Description

name	Name of the translation profile. Maximum length is 31 alphanumeric characters.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

After translation rules are defined, they are grouped into profiles. The profiles collect a set of rules that, taken together, translate the called, calling, and redirected numbers in specific ways. Up to 1000 profiles can be defined. Each profile must have a unique name.

These profiles are referenced by trunk groups, dial peers, source IP groups, voice ports, and interfaces for handling call translations.

Examples

The following example initiates translation profile "westcoast" for voice calls. The profile uses translation rules 1, 2, and 3 for various types of calls.

```
Router(config)# voice translation-profile westcoast
Router(cfg-translation-profile)# translate calling 2
Router(cfg-translation-profile)# translate called 1
Router(cfg-translation-profile)# translate redirect-called 3
```

Related Commands

Command	Description
rule (voice translation-rule)	Defines call translation criteria.
show voice translation-profile	Displays one or more translation profiles.
translate (translation profiles)	Associates a translation rule with a voice translation profile.

voice translation-rule

To define a translation rule for voice calls, use the **voice translation-rule** command in global configuration mode. To delete the translation rule, use the **no** form of this command.

voice translation-rule *number*

no voice translation-rule *number*

Syntax Description

number	Unique identifier for the translation rule. Range is from1 to 2147483647. There is no default.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the **voice translation-rule** command to create the definition of a translation rule. Each definition includes up to 15 rules that include SED-like expressions for processing the call translation. A maximum of 128 translation rules are supported.

These translation rules are grouped into profiles that are referenced by trunk groups, dial peers, source IP groups, voice ports, and interfaces.

Examples

The following example initiates translation rule 150, Which includes two rules:

```
Router(config)# voice translation-rule 150
Router(cfg-translation-rule)# rule 1 reject /^408\(.(\)/
Router(cfg-translation-rule)# rule 2 /\(^...\)853\(...\)/ /\1525\2/
```

Related Commands

Command	Description
rule (voice translation-rule)	Defines the matching, replacement, and rejection patterns for a translation rule.
show voice translation-rule	Displays the configuration of a translation rule.

voice vad-time

To change the minimum silence detection time for voice activity detection (VAD), use the **voice vad-time** command in global configuration mode. To reset to the default, use the **no** form of this command.

voice vad-time *milliseconds*

no voice vad-time

Syntax Description

milliseconds	Waiting period, in milliseconds, before silence detection and suppression of voice-packet transmission. Range is from 250 to 65536. Default is 250.

Defaults

250 milliseconds

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.1(2)T	This command was integrated into this release.

Usage Guidelines

This command affects all voice ports on a router or concentrator, but it does not affect calls already in progress.

You can use this command in transparent common-channel signaling (CCS) applications in which you want VAD to activate when the voice channel is idle, but not during active calls. With a longer silence detection delay, VAD reacts to the silence of an idle voice channel, but not to pauses in conversation.

This command does not affect voice codecs that have ITU-standardized built-in VAD features—for example, G.729B, G.729AB, G.723.1A. The VAD behavior and parameters of these codecs are defined exclusively by the applicable ITU standard.

Examples

The following example configures a 20-second delay before VAD silence detection is enabled:

```
voice vad-time 20000
```

Related Commands

Command	Description
vad (dial peer)	Enables voice activity detection on a network dial peer.

voice-card

To enter the voice-card configuration mode and configure a voice card, use the **voice-card** command in global configuration mode.

voice-card *slot*

Syntax Description

slot	Slot number for the card to be configured. The following platform-specific numbering schemes apply: • Cisco 2600 series and Cisco 2600XM – 0 is the Advanced Integration Module (AIM) slot. – 1 is the network module slot . • Cisco 3600 series – 1to 6 are network-module slots. • Cisco 3660 – 7 is AIM slot 0. – 8 is AIM slot 1. • Cisco MC3810 with one or two high-performance voice-compression modules (HCMs) installed – 0 applies to the entire chassis.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XK	The command was introduced on the Cisco 2600 series and Cisco 3600 series.
12.0(7)T	This command was integrated into this release.
12.0(7)XK	This command was implemented on the Cisco MC3810.
12.1(2)T	This command was integrated into this release.
12.2(2)XB	Values for the *slot* argument were updated to include AIMs.
12.2(8)T	This command was integrated into this release.
12.2(13)T	This command was implemented on the Cisco 1700 series, Cisco 2600XM, Cisco 3700 series, Cisco 7200 series, Cisco 7500 series, Cisco ICS7750, Cisco MC3810, and Cisco VG200.
12.2(15)T	This command was integrated into this release.

Usage Guidelines

Voice-card configuration mode is used for commands that configure the use of digital signal processing (DSP) resources, such as codec complexity and DSPs. DSP resources can be found in digital T1/E1 packet voice trunk network modules on Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series, and on high-performance compression modules on Cisco MC3810 multiservice access concentrators.

Codec complexity is configured in voice-card configuration mode and has the following platform-specific usage guidelines:

- On Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, and Cisco 3745, the *slot* argument corresponds to the physical chassis slot of the network module that has DSP resources to be configured.
- On the Cisco MC3810, the *slot* argument is always 0, and the changes that are made in voice-card mode apply to the entire Cisco MC3810. On the Cisco MC3810, the **voice-card** command is available only if the chassis is equipped with one or two HCMs.

DSP resource sharing is also configured in voice-card configuration mode. On the Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, and Cisco 3745 under specific circumstances, configuration of the **dspfarm** command enters DSP resources on a network module or AIM into a DSP resource pool. Those DSP resources are then available to process voice traffic on a different network module or voice/WAN interface card (VWIC). See the **dspfarm (voice-card)** command reference for more information about DSP resource sharing.

Note When running high-complexity images, the system can only process up to 16 voice channels. Those 16 time slots need to be within a contiguous range (timeslot maximum (TSmax) minus timeslot minimum (TSmin) is less than or equal to 16, where TSmax and TSmin are the maximum DS0 and minimum DS0 configured for voice).

This command does not have a **no** form.

Examples

The following example enters voice-card configuration mode to configure resources on the network module in slot 1 on a Cisco 2600 series or Cisco 3600 series router:

```
voice-card 1
```

The following example enters voice-card configuration mode on a Cisco MC3810:

```
voice-card 0
```

The following example shows how to enter voice-card configuration mode and load high-complexity DSP firmware on voice-card 0. The **dspfarm** command enters the DSP resources on the AIM specified in the **voice-card** command into the DSP resource pool.

```
voice-card 0
 codec complexity high
 dspfarm
```

Related Commands

Command	Description
codec complexity	Matches the DSP complexity packaging to the codecs to be supported.
dspfarm (voice-card)	Adds the specified voice card to those participating in a DSP resource pool.

voice-class sip rel1xx

To enable all Session Initiation Protocol (SIP) provisional responses (other than 100 Trying) to be sent reliably to the remote SIP endpoint, use the **voice-class sip rel1xx** command in dial-peer configuration mode. To reset to the default, use the **no** form of this command.

voice-class sip rel1xx {**supported** *value* | **require** *value* | **system** | **disable**}

no sip rel1xx

Syntax Description

supported *value*	Supports reliable provisional responses. The *value* argument may have any value, as long as both the user-agent client (UAC) and user-agent server (UAS) configure it the same.
require *value*	Requires reliable provisional responses. The *value* argument may have any value, as long as both the UAC and UAS configure it the same.
system	Uses the value configured in voice service mode. This is the default.
disable	Disables the use of reliable provisional responses.

Defaults

system

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into this release. The following were not supported: the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

There are two ways to configure reliable provisional responses:

- Dial-peer mode. You can configure reliable provisional responses for the specific dial peer only by using the **voice-class sip rel1xx** command.
- SIP mode. You can configure reliable provisional responses globally by using the **rel1xx** command.

The use of resource reservation with SIP requires that the reliable provisional feature for SIP be enabled either at the VoIP dial-peer level or globally on the router.

This command applies to the dial peer under which it is used or points to the global configuration for reliable provisional responses. If the command is used with the **supported** keyword, the SIP gateway uses the Supported header in outgoing SIP INVITE requests. If it is used with the **require** keyword, the gateway uses the Required header.

This command, in dial-peer configuration mode, takes precedence over the **rel1xx** command in global configuration mode with one exception: If this command is used with the **system** keyword, the gateway uses what was configured under the **rel1xx** command in global configuration mode.

Examples

The following example shows how to use this command on either an originating or a terminating SIP gateway:

- On an originating gateway, all outgoing SIP INVITE requests matching this dial peer contain the Supported header where *value* is 100rel.
- On a terminating gateway, all received SIP INVITE requests matching this dial peer support reliable provisional responses.

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip rel1xx supported 100rel
```

Related Commands

Command	Description
rel1xx	Provides provisional responses for calls on all VoIP calls.

voice-class sip url

To configure URLs to either the Session Initiation Protocol (SIP) or telephone (TEL) format for your dial-peer SIP calls, use the **voice-class sip url** command in dial-peer configuration mode. To reset to the default, use the **no** form of this command.

voice-class sip url {**sip** | **tel** | **system**}

no voice-class sip url

Syntax Description

sip	Generates URLs in the SIP format for calls on a dial-peer basis.
tel	Generates URLs in the TEL format for calls on a dial-peer basis.
system	Uses the system value. This is the default.

Defaults

system

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into this release. The following were not supported: the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, and Cisco AS5400.

Usage Guidelines

This command affects only user-agent clients (UACs), because it causes the use of a TEL or SIP URL in the request line of outgoing SIP INVITE requests. SIP URLs indicate the originator, recipient, and destination of the SIP request; TEL URLs indicate voice-call connections.

The **voice-class sip url** command, in dial-peer configuration mode, takes precedence over the **url** command in SIP global-configuration mode. However, if the **voice-class sip url** command is used with the **system** keyword, the gateway uses what was globally configured under the **url** command.

Examples

The following example shows how to set up the **voice-class sip url** command to generate URLs in the TEL format:

```
Router(config)# dial-peer voice 102 voip
Router(config-dial-peer)# voice-class sip url tel
```

Related Commands

Command	Description
sip url	Generates URLs in the SIP or TEL format in VoIP configuration mode.

voice-encap

This command was added in Cisco IOS Release 11.3(1)MA on Cisco MC3810. This command is not supported in Cisco IOS Release 12.2.

voice-group

This command was added in Cisco IOS Release 11.3(1)MA for Cisco MC3810. This command is not supported in Cisco IOS Release 12.2.

voicemail (cm-fallback)

To configure the telephone number that is speed-dialed when the messages button on a Cisco IP phone is pressed, use the **voicemail** command in call-manager-fallback configuration mode. To disable the messages button, use the **no** form of this command.

voicemail *phone-number*

no voicemail

Syntax Description

phone-number	Phone number that is configured as a speed-dial number for retrieving messages.

Defaults

No phone number is configured, and the messages button is ineffective.

Command Modes

Call-manager-fallback configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3.
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 configures the telephone number that is speed-dialed when the message button on a Cisco IP phone is pressed. The same voicemail telephone number is configured for all Cisco IP phones connected to the router.

Examples

The following example sets the phone number 4085551000 as the speed-dial number that is dialed to retrieve messages when the messages button is pressed:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# voicemail 914085551000
```

The number 914085551000 is called when the Cisco IP phone messages button is pressed to retrieve messages.

Related Commands

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

voicemail (telephony-service)

To configure the telephone number that is speed-dialed when the messages button on a Cisco IP phone is pressed, use the **voicemail** command in telephony-service configuration mode. To disable the messages button, use the **no** form of this command.

voicemail *phone-number*

no voicemail

Syntax Description

phone-number	Phone number that is configured as a speed-dial number for retrieving messages.

Defaults

No phone number is configured, and the messages button is ineffective.

Command Modes

Telephony-service configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the Cisco 2600, Cisco 3600, and Cisco IAD2420.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was 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 configures the telephone number that is speed-dialed when the messages button on a Cisco IP phone is pressed. The same telephone number is configured for voice mail for all Cisco IP phones connected to the router.

Examples

The following example sets the phone number 914085551000 as the speed-dial number that is dialed to retrieve messages when the messages button is pressed:

```
Router(config)# telephony-service
Router(config-telephony-service)# voicemail 914085551000
```

The number 914085551000 is called when the Cisco IP phone messages button is pressed to retrieve messages.

Related Commands

Command	Description
telephony-service	Enables Cisco IOS Telephony Service and enters telephony-service configuration mode.
vm-device-id (ephone)	Defines the voice-mail ID string.

voice-port

To enter voice-port configuration mode, use the **voice-port** command in global configuration mode.

Cisco 1750 and Cisco 1751

voice-port *slot-number***/***port*

Cisco 2600, Cisco 3600 Series and Cisco 7200 Series

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

Cisco 2600 and Cisco 3600 Series with a High-Density Analog Network Module (NM-HDA)

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

Cisco AS5300

voice-port *controller-number***:D**

Cisco AS5800

voice-port {*shelf***/***slot***/***port***:D** | *shelf***/***slot***/***parent***:***port***:D**}

Cisco MC3810

voice-port *slot***/***port*

Syntax Description

Cisco 1750 and Cisco 1751

slot-number	Number of the slot in the router in which the voice interface card (VIC) is installed. Range is from 0 to 2, depending on the slot in which it is installed.
port	Voice port number. Range is from 0 to 1.

Cisco 2600, Cisco 3600 Series and Cisco 7200 Series

slot-number	Number of the slot in the router in which the VIC is installed. Range is from 0 to 3, depending on the slot in which it is installed.
subunit-number	Subunit on the VIC in which the voice port is located. Range is from 0 to 1.
port	Voice port number. Range is from 0 to 1.
slot	Router location in which the voice port adapter is installed. Range is from from 0 to 3.
*port***:**	VIC location. Range is from 0 to 3.
ds0-group-no	Defined DS0 group number. Each such number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

Cisco AS5300:

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

Cisco AS5800:

shelf	T1 or E1 controller on the T1 card, or the T1 controller on the T3 card. Range is from 0 to 9999.
slot	T1 or E1 controller on the T1 card, or the T1 controller on the T3 card. Range is from 0 to 11.
port	Voice port number. • T1 or E1 controller on the T1 card range is from 0 to 11. • T1 controller on the T3 card range is from 1 to 28.
:*port*	Value for the *parent* argument. Valid entry is 0.
:D	D channel associated with ISDN PRI.

Cisco MC3810

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

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(3)T	This command was implemented on the Cisco 2600 series.
12.0(3)T	This command was implemented on the Cisco AS5300.
12.0(7)T	This command was implemented on the Cisco AS5800, Cisco 7200 series, and Cisco 1750. Arguments were added for the Cisco 2600 series and Cisco 3600 series.

Release	Modification
12.2(8)T	This command was implemented on the Cisco 1751 and Cisco 1760. The command was modified to accommodate the additional ports of the NM-HDA on the Cisco 2600 series, Cisco 3640, and Cisco 3660.
12.2(2)XN	Support for enhanced MGCP voice gateway interoperability was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(11)T	This command was integrated into Cisco CallManager Version 3.2 and implemented on the Cisco IAD2420 series.
12.2(13)T	This command was integrated into this release. The following was not supported: the extended echo canceller (EC) feature on the Cisco AS5300 and Cisco AS5800.

Usage Guidelines

Use the **voice-port** global configuration command to switch to voice-port configuration mode from global configuration mode. Use the **exit** command to exit voice-port configuration mode and return to global configuration mode.

Note This command does not support the extended echo canceller (EC) feature on the Cisco AS5300 or the Cisco AS5800.

Examples

The following example accesses voice-port configuration mode for port 0, located on subunit 0 on a VIC installed in slot 1 of a Cisco 3600 series router:

```
voice-port 1/0/0
```

The following example accesses voice-port configuration mode for digital voice port 24 on a Cisco MC3810 that has a digital voice module (DVM) installed:

```
voice-port 1/24
```

The following example accesses voice-port configuration mode for a Cisco AS5300:

```
voice-port 1:D
```

The following example accesses voice-port configuration mode for a Cisco AS5800 (T1 card):

```
voice-port 1/0/0:D
```

The following example accesses voice-port configuration mode for a Cisco AS5800 (T3 card):

```
voice-port 1/0/0:1:D
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.

voice-port (MGCP profile)

The **voice-port** (MGCP profile) command is replaced by the **port** (MGCP profile) command in Cisco IOS Release 12.2(8)T. See the **port** (MGCP profile) command for more information.

voice-port busyout

To place all voice ports associated with a serial or ATM interface into a busyout state, use the **voice-port busyout** command in interface configuration mode. To remove the busyout state on the voice ports associated with this interface, use the **no** form of this command.

voice-port busyout

no voice-port busyout

Syntax Description

This command has no arguments or keywords.

Defaults

The voice ports on the interface are not in busyout state.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco MC3810.

Usage Guidelines

This command busies out all voice ports associated with the interface, except any voice ports configured to busy out under specific conditions using the **busyout monitor** and **busyout seize** commands.

Examples

The following example places the voice ports associated with serial interface 1 into busyout state:

```
interface serial 1
 voice-port busyout
```

The following example places the voice ports associated with ATM interface 0 into busyout state:

```
interface atm 0
 voice-port busyout
```

Related Commands

Command	Description
busyout forced	Forces a voice port on the Cisco MC3810 into the busyout state.
busyout monitor	Places a voice port on the Cisco MC3810 into the busyout monitor state.
busyout seize	Changes the busyout action for an FXO or FXS voice port.
show voice busyout	Displays information about the voice busyout state on the Cisco MC3810.

voip-incoming translation-profile

To specify a translation profile for all incoming VoIP calls, use the **voip-incoming translation-profile** command in global configuration mode. To delete the profile, use the **no** form of this command.

voip-incoming translation-profile *name*

no voip-incoming translation-profile *name*

Syntax Description

name	Name of the translation profile.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the **voip-incoming translation-profile** command to globally assign a translation profile for all incoming VoIP calls. The translation profile was previously defined using the **voice translation-profile** command. The **voip-incoming translation-profile** command does not require additional steps to complete its definition.

If an H.323 call comes in and the call is associated with a source IP group that is defined with a translation profile, the source IP group translation profile overrides the global translation profile.

Examples

The following example assigns the translation profile named "global-definition" to all incoming VoIP calls:

```
Router(config)# voip-incoming translation-profile global-definition
```

Related Commands

Command	Description
show voice translation-profile	Displays the configurations for all voice translation profiles.
test voice translation-rule	Tests the voice translation rule definition.
voice translation-profile	Initiates a translation profile definition.

voip-incoming translation-rule

To set the incoming translation rule for calls that originate from H.323-compatible clients, use the **voip-incoming translation-rule** command in global configuration mode. To disable the incoming translation rule, use the **no** form of this command.

voip-incoming translation-rule *tag* {**calling-number** | **called-number**}

no voip-incoming translation-rule *tag* {**calling-number** | **called-number**}

Syntax Description

tag	Tag number by which the rule set is referenced. This is an arbitrarily chosen number. Range is from 1 to 2147483647. There is no default value.
calling-number	Automatic number identification (ANI) number or the number of the calling party.
called-number	Dial Number Information Service (DNIS) number or the number of the called party.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for VoIP on the Cisco AS5300.
12.0(7)XK	This command was implemented for VoIP on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.1(1)T	This command was implemented for VoIP on the Cisco 1750, Cisco AS5300, Cisco 7200, and Cisco 7500.
12.1(2)T	This command was implemented for VoIP on the Cisco MC3810.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into this release.

Usage Guidelines

With this command, all IP-based calls are captured and handled, depending on either the calling number or the called number to the specified tag name.

Examples

The following example identifies the rule set for calls that originate from H.323-compatible clients:

```
Router(config)# voip-incoming translation-rule 5 called-number
```

Related Commands

Command	Description
numbering-type	Matches one number type for a dial-peer call leg.
rule	Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls.
show translation-rule	Displays the contents of all the rules that have been configured for a specific translation name.
test translation-rule	Tests the execution of the translation rules on a specific name-tag.
translate	Applies a translation rule to a calling party number or a called party number for incoming calls.
translate-outgoing	Applies a translation rule to a calling party number or a called party number for outgoing calls.
translation-rule	Creates a translation name and enters translation-rule configuration mode.

volume

To set the receiver volume level for a POTS port on a router, use the **volume** command in dial-peer voice configuration mode. To reset to the default, use the **no** form of this command.

volume *number*

no volume *number*

Syntax Description

number	Decibels (dB) of gain. Range is as follows: • 1: –11.99 dB • 2: –9.7dB • 3: –7.7dB • 4: –5.7dB • 5: –3.7dB Default is 3 (–7.7 dB gain).

Defaults

3 (–7.7 dB gain)

Command Modes

Dial-peer voice configuration

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco 803, Cisco 804, and Cisco 813.

Usage Guidelines

Set the **volume** command for each POTS port separately. Setting the volume level affects only the port for which it has been set.

Note Only the receiver volume is set with this command.

Use the **show pots volume** command to check the volume status and level.

Examples

The following example shows a volume level of 4 for POTS port 1 and a volume level of 2 for POTS port 2.

```
dial-peer voice 1 pots
 destination-pattern 5551111
 port 1
 no call-waiting
 ring 0
 volume 4
```

```
dial-peer voice 2 pots
 destination-pattern 5552222
 port 2
 no call-waiting
 ring 0
 volume 2
```

Related Commands

Command	Description
show pots volume	Shows the receiver volume configured for each POTS port on a router.

web admin customer

To define a username and password for a Cisco IOS Telephony System (ITS) customer administrator, use the **web admin customer** command in telephony-service configuration mode. To disable a customer administrator login, use the **no** form of this command.

web admin customer name *username* {**password** *string* | **secret** {**0** | **5**} *string*}

no web admin customer

Syntax Description

name *username*	Username for the customer administrator. Default is Customer.
password *string*	Password for the customer administrator. Default is no password.
secret {**0** | **5**} *string*	Secret password and whether or not it is encrypted. Keywords are as follows: • **0**—Password that follows is not encrypted. • **5**—Password that follows is encrypted.

Defaults

A customer administrator named Customer with no password is defined.

Command Modes

Telephony-service

Command History

Release	Modification
12.2(11)YT	This command was introduced.
12.2(15)T	This command was integrated into this release.

Usage Guidelines

Use this command with Cisco IOS Telephony Service (ITS) V2.1 or a later version.

Examples

The following example defines a customer administrator named user22 whose password is pw567890:

```
Router(config)# telephony-service
Router(config-telephony-service)# web admin customer name user22 password pw567890
```

Related Commands

Command	Description
telephony-service	Enables Cisco ITS and enters telephony-service configuration mode.
web customize load	Loads and parses an eXtensible Markup Language (XML) file in router Flash memory to customize a graphical user interface (GUI) for a customer administrator using Cisco ITS.

web admin system

To define a username and password for a Cisco IOS Telephony Service (ITS) system administrator, use the **web admin system** command in telephony-service configuration mode. To disable a customer administrator login, use the **no** form of this command.

web admin system name *username* {**password** *string* | **secret** {**0** | **5**} *string*}

no web admin system

Syntax Description

name *username*	Login name for the system administrator. Default is Admin.
password *string*	Character string for login authentication, stored in the running configuration as plain text. Default is no password.
secret {**0** | **5**} *string*	Character string for login authentication, stored in the running configuration as encrypted using MD5, and whether or not it is encripted. Keywords are as follows: • **0**—Password that follows is not encrypted. • **5**—Password that follows is encrypted.

Defaults

A system administrator named Admin with no password is defined.

Command Modes

Telephony-service

Command History

Release	Modification
12.2(11)YT	This command was introduced.
12.2(15)T	This command was integrated into this release.

Usage Guidelines

Use this command with Cisco ITS V2.1 or a later version.

You can encrypt the system administrator password with MD5 by using the **secret 0** keyword pair before entering a plain-text password string. An encrypted version of the string is saved in the running configuration, as shown in the following example. Note that the digit 5 appears in this line to indicate that the password that follows is shown in its encrypted version.

```
web admin system name jsmith secret 5 $1$TCyK$OU/NSQ/VtAU2ibHdi8Uau
```

Examples

The following example establishes a system administrator named user1 whose password is pw234567:

```
Router(config)# telephony-service
Router(config-telephony-service)# web admin system name user1 password pw234567
```

Related Commands

Command	Description
telephony-service	Enables Cisco ITS and enters telephony-service configuration mode.

web customize load

To load and parse an eXtensible Markup Language (XML) file in router Flash memory to customize a graphical user interface (GUI) for a customer administrator using Cisco IOS Telephony Service (ITS), use the **web customize load** command in telephony-service configuration mode. To disable the customized GUI and fall back to the system administrator GUI, use the **no** form of this command.

web customize load *filename*

no web customize load

Syntax Description

filename	XML file in flash memory that is to be loaded and parsed. This file defines the customer administrator GUI.

Defaults

The standard system administrator GUI is used.

Command Modes

Telephony-service configuration

Command History

Release	Modification
12.2(11)YT	This command was introduced.
12.2(15)T	This command was integrated into this release.

Usage Guidelines

Use this command with Cisco ITS V2.1 or a later version.

Examples

The following example specifies a file named cust_admin_gui.xml as the file that defines the GUI for ITS customer administrators:

```
Router(config)# telephony-service
Router(config-telephony-service)# web customize load cust_admin_gui.xml
```

Related Commands

Command	Description
telephony-service	Enables Cisco ITS and enters telephony-service configuration mode.