vad (dial peer) through voice-class sip encap clear-channel

vad (dial peer)

To enable voice activity detection (VAD) for calls using a specific 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 to -62 dBm. Available only when session protocol multicast is configured.

Command Default

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 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.
Cisco IOS XE Bengaluru 17.6.1a	Introduced support for YANG models.

Usage Guidelines

Use thiscommand 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.

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 (SPA-DSP)

To enable or disable voice activity detection (vad) settings configured locally irrespective of the external vad settings, use the **vad**command in config dspfarm profile mode.

vad {**on** | **off**} **override**

Syntax Description

on	Enables the local vad settings irrespective of the external vad settings.
off	Disables the local vad settins irrespective of the external vad settings.
override	Overrides the external vad settings with local vad configuration details.

Command Default

By default, VAD is enabled.

Command Modes

DSP Farm Profile Configuration Mode (config-dspfarm-profile)

Command History

Release	Modification
Cisco IOS XE Release 3.2S	This command was introduced.

Usage Guidelines

Use this command to enable voice activity detection locally irrespective of external VAD settings. 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 disable VAD, voice data is continuously sent to the IP backbone.

Examples

The following example enables VAD and overrides external vad settings with local vad settings:

```
Router(config)# dspfarm profile 1
Router(config-dspfarm-profile)# vad on override
Router(config-dspfarm-profile)# do show running-config
!!!
dspfarm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 588
 associate application SBC
 vad on override
!
```

The following example disables local vad settings and overrides external vad setting configuration:

```
Router(config)# dspfarm profile 1
Router(config-dspfarm-profile)# vad off override
Router(config-dspfarm-profile)# do show running-config
!!!
dspfarm profile 1 transcode
 codec g711ulaw
```

```
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 588
 associate application SBC
 vad off override
!
```

Related Commands

Command	Description
dsp services dspfarm	Enables the DSP-farm services.
dspfarm profile	Enters the DSP farm profile configuration mode, and defines a profile for the DSP farm services.
show dspfarm (SPA-DSP)	Displays DSP farm service information, such as operational status and DSP resource allocation for transcoding.

vbd-playout-delay

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

vbd-playout-delay {**maximum** *milliseconds* | **minimum** *milliseconds* | **mode** {**fixed** [**no-timestamps**] | **passthrough**} | **nominal** *milliseconds*}
no vbd-playout-delay

Syntax Description

maximum	Sets the maximum playout buffer delay, in milliseconds (ms). Range: 40 to 1000. Default: 1000.
milliseconds	Delay time, in milliseconds (ms).
minimum	Sets the minimum playout buffer delay, in ms. Range: 10 to 40. Default: 40.
mode	Configures voice-band-detection playout buffer adaptation mode.
fixed	Sets the jitter buffer to a constant delay.
no-timestamps	(Optional) Fixes the jitter buffer at a constant delay without time stamps.
passthrough	Sets the jitter buffer passthrough mode for clock compensation.
nominal	Sets the nominal playout buffer delay, in ms. Range:10 to 1000. Default: 60.

Command Default

The voice-band-detection playout-delay buffer is disabled.

Command Modes

Voice service session configuration (conf-voi-serv-sess)

Command History

Release	Modification
12.2(8)T	This command was introduced.
12.4(24)T	This command was modified. • The minimum time range value was changed from 4 to 1700 ms to a range of 10 to 40 ms. The default value 4 was increased to 40 ms. • The maximum time value was decreased from 1700 to 1000 ms and the default of 200 was increased to 1000 ms. • The nominal time range value was changed from 0 to 1500 ms to a range of 10 to 1000 ms. The default value of 100 was decreased to 60 ms.
12.4(24)T6	This command was modified. The **no-timestamps** keyword was added and **passthrough** keyword usage guidelines were clarified.

Usage Guidelines

Use thiscommand to set the playout jitter buffer. When a voice band is detected, the call uses the 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. The **no-timestamps** keyword sets the jitter buffer at a constant delay without reading time stamps.

Note The **passthough** keyword is a special mode used to handle clock drifting properly. We recommend this keyword only when instructed by your Cisco representative.

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
```

The following example configures AAL2 voice-band-detection playout-delay adaptation mode and sets the mode at a constant delay without timestamps:

```
voice service voatm
 session protocol aal2
  vbd-playout-delay mode fixed no-timestamps
```

The following example sets the nominal AAL2 voice-band-detection playout-delay buffer to 12 ms:

```
voice service voatm
 session protocol aal2
  vbd-playout-delay nominal 12
```

The following example sets the AAL2 voice-band-detection playout-buffer delay to a maximum of 55 ms:

```
voice service voatm
 session protocol aal2
  vbd-playout-delay maximum 55
```

The following example sets the AAL2 voice-band-detection playout-buffer delay to a minimum of 22 ms:

```
voice service voatm
 session protocol aal2
  vbd-playout-delay minimum 22
```

The following sample output shows the vdb-playout-delay being verified in the running configuration output:

```
Router(conf-voi-serv-sess)#do show run | sec voice service voatm
voice service voatm
 !
 session protocol aal2
  vbd-playout-delay minimum 22
```

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 kilobytes per second (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.

Command Default

No real-time VBR settings are configured

Command Modes

ATM Bundle-vc configuration for ATM VC bundle members
ATM PVP configuration for an ATM PVP
Interface-ATM-VC configuration for an ATM permanent virtual connection (PVC) or switched virtual circuit (SVC)
VC-class configuration for a virtual circuit (VC) class

Command History

Release	Modification
12.0	This command was introduced on the Cisco MC3810.
12.1(5)XM	This command was implemented on Cisco 3600 series routers and modified to support Simple Gateway Control Protocol (SGCP) and Media Gateway Control Protocol (MGCP).
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T.
12.2(11)T	This command was implemented on the Cisco AS5300 and Cisco AS5850.
Cisco IOS XE Release 2.3	This command was made available in ATM PVP configuration mode.

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 value.

Command Default

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 Cisco IOS Release 12.2(2)T.
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.

video codec (dial peer)

To assign a video codec to a VoIP dial peer, use the **video codec** command in dial peer configuration mode. To remove a video codec, use the **no** form of this command.

video codec {**h261** | **h263** | **h263+** | **h264**}
no video codec

Syntax Description

h261	Video codec H.261
h263	Video codec H.263
h263+	Video codec H.263+
h264	Video codec H.264

Command Default

No video codec is configured.

Command Modes

Dial peer configuration

Command History

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

Usage Guidelines

Use this command to configure a video codec for a VoIP dial peer. If no video codec is configured, the default is transparent codec operation between the endpoints.

Examples

The following example shows configuration for video codec H.263+ on VoIP dial peer 30:

```
dial-peer voice 30 voip
 video codec h263+
```

Related Commands

Command	Description
video codec (voice-class)	Specifies a video codec for a voice class.

video codec (voice class)

To specify a video codec for a voice class, use the **video codec**command in voice class configuration mode. To remove the video codec, use the **no** form of this command.

video codec {**h261** | **h263** | **h263+** | **h264**}
no video codec {**h261** | **h263** | **h263+** | **h264**}

Syntax Description

h261	Apply this preference to video codec H.261
h263	Apply this preference to video codec H.263
h263+	Apply this preference to video codec H.263+
h264	Apply this preference to video codec H.264

Command Default

No video codec is configured.

Command Modes

Voice class configuration

Command History

Release	**Modification**
12.4(11)T	This command was introduced.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models.

Usage Guidelines

Use this command to specify one or more video codecs for a voice class.

Examples

The following example shows configuration for voice class codec 10 with two audio codec preferences and three video codec preferences:

```
voice class codec 10
 codec preference 1 g711alaw
 codec preference 2 g722
 video codec h261
 video codec h263
 video codec h264
 video codec mpeg4
```

Related Commands

Command	**Description**
video codec (dial peer)	Specifies a video codec for a VoIP dial peer.

video screening

To enable transcoding and transsizing between two call legs when configuring SIP, use the **video screening** command in voice service SIP configuration mode or voice class tenant configuration mode. To disable transcoding and transsizing, use **no** form of this command.

video screening system
no video screening system

Syntax Description

system	Specifies that transcoding and transsizing use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

Command Default

Video screening is disabled.

Command Modes

Voice service SIP configuration.

Voice class tenant configuration (config-class)

Command History

Release	Modification
15.1(4)M	The command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: **system**. This command is now available under voice class tenants.

Usage Guidelines

Use this command to enable conversion of video streams if there is a mismatch between two call legs.

Examples

The following example enters the voice-card configuration mode and enables video screening:

```
Router(config)# voice service voip
Router(config-voicecard)# sip
Router((conf-serv-sip)# video screening
```

The following example enters the voice-card configuration mode and enables video screening in voice class tenant configuration mode:

```
Router(conf-class)# video screening system
```

Related Commands

Command	Description
codec profile	Defines the video capabilities needed for video endpoints.
video codec	Assigns a video codec to a VoIP dial peer.

violation

To specify the action that needs to be performed on any violation in the Differentiated Services Code Point (DSCP) policy, use the **violation** command in voice class configuration mode. To disable the configuration, use the **no** form of this command.

violation *number* **action** {**disconnect** | **ignore**} [{**no-syslog**}]
no violation *number* **action** {**disconnect** | **ignore**} [{**no-syslog**}]

Syntax Description

number	Number of violations after which the required action needs to be taken. The range is from 1 to 200000. The default value is 20.
action	Specifies that an action must be performed after the specified number of violations.
disconnect	Disconnects the call after the specified number of violations is exceeded.
ignore	Specifies that no action should be taken after the specified number of violations is exceeded.
no-syslog	(Optional) Specifies not to print messages to the system log when violations occur.

Command Default

No actions are specified against any violation.

Command Modes

Voice class configuration (config-class)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

You can use the **violation** command to specify the action that needs to be performed on any violation in the DSCP policy. A system log is created by default. You can configure the **no-syslog** keyword to disable the Cisco Unified Border Element (Cisco UBE) from generating system logs on DSCP policy violation.

Configure a high value for DSCP violations. If you configure a low value such as 5, action will be performed on the call after every five violations and system logs will be generated frequently.

The “100 - Invalid information element contents [Q.850]” message is displayed in the system log when a call is disconnected because of a DSCP policy violation. The cause for disconnecting the call is propagated only to the call leg causing the violation. For example, if the outgoing call leg of a Session Initiation Protocol (SIP)-to-SIP call violates the DSCP policy and the number of violations exceeds the configured number, this call is disconnected with the cause of 100 (Invalid information element contents [Q.850]) to the outgoing call leg and cause 16 (Normal Call Cleaning) to the incoming call leg.

Examples

The following example shows how to configure a router to print to the system log and disconnect the call if a call exceeds 20,000 violations:

```
Router> enable
Router# configure terminal
Router(config)# voice class dscp-profile 1
Router(config-class)# violation 20000 action disconnect
```

Related Commands

Command	Description
dscp media	Specifies the RPH to DSCP mapping.
voice class dscp-profile	Configures the DSCP profile.

violation (media profile)

To specify the action that needs to be performed on any violation in the media bandwidth policy, use the **violation** command in media profile configuration mode. To disable the configuration, use the **no** form of this command.

violation *number* **action** {**disconnect** | **drop** | **ignore**} [{**no-syslog**}]
no violation *number* **action** {**disconnect** | **drop** | **ignore**} [{**no-syslog**}]

Syntax Description

number	Number of violations after which the required action needs to be taken. The range is from 1 to 200000. The default value is 20.
action	Specifies that an action must be performed after the specified number of violations.
disconnect	Disconnects the call after the specified number of violations is exceeded.
drop	Drops the call after the specified number of violations is exceeded.
ignore	Specifies that no action should be taken after the specified number of violations is exceeded.
no-syslog	(Optional) Specifies not to print messages to the system log when violations occur.

Command Default

No actions are specified against any violation.

Command Modes

Media profile configuration (cfg-mediaprofile)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

You can use the **violation** command to specify the action that needs to be performed on any violation in the media bandwidth policy. A system log is created by default. You can configure the **no-syslog** keyword to disable the Cisco Unified Border Element (Cisco UBE) from generating system logs on DSCP policy violation.

Configure a high value for DSCP violations. If you configure a low value such as 5, action will be performed on the call after every five violations and system logs will be generated frequently.

Examples

The following example shows how to configure a router to print the system log and disconnect the call if a call exceeds 20,000 violations:

```
Router> enable
Router# configure terminal
Router(config)# media profile police 1
Router(cfg-mediaprofile)# violation 20000 action drop
```

Related Commands

Command	Description
media profile police	Configures the media policing profile.
overhead	Configures the overhead bandwidth percentage above the negotiated bandwidth.

vmwi

To enable DC voltage or FSK visual message-waiting indictator (VMWI) on a Cisco VG224 onboard analog FXS voice port, use the **vmwi** command in voice-port configuration mode. To reset VMWI to default, use the **no** form of this command.

vmwi {**dc-voltage** | **fsk**}
no vmwi

Syntax Description

dc-voltage	DC voltage VMWI is enabled on this FXS port.
fsk	FSK VMWI is enabled on this FXS port. Default.

Command Default

FSK VMWI is enabled.

Command Modes

Voice-port configuration (config-voiceport)

Command History

Release	Modification
12.4(20)YA	This command was introduced.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.

Usage Guidelines

This command with the **dc-voltage** keyword enables the message-waiting lamp to flash on an analog phone that requires DC voltage to activate a visual indicator.

This command with the **fsk** keyword enables the message-waiting lamp to flash on an analog phone that requires an FSK message to activate a visual indicator.

DC Voltage VMWI is supported for the SCCP telephony control (STC) application only. For all other applications, such as MGCP, FSK will be used even if you configure the **vmwi dc-voltage** command on the voice gateway.

Examples

The example shows how to enable DC Voltage VMWI on port 2/0 on a Cisco VG224.

```
Router(config)#voice-port 2/0
Router(config-voiceport)#vmwi dc-voltage
Router(config-voiceport)#end
```

Related Commands

Command	Description
stcapp	Enables basic SCCP call-control features for FXS analog ports on Cisco IOS voice gateways

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 Release Before 12.0(7)XK and Release 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*

FRF.11 Trunk Calls
vofr [**data** *cid*] [**call-control** *cid*]
no vofr [**data** *cid*] [**call-control** *cid*]

Syntax Description

data	(Required for Cisco-trunk permanent calls. Optional for switched calls.) Selects a subchannel (CID) for data other than the default subchannel, which is 4.
cid	(Optional) Specifies the subchannel to be used for data. Range is from 4 to 255. The default is 4. If **data** is specified, enter a valid CID.
call-control	(Optional) Reserves a subchannel for call-control signaling.
cisco	(Optional) Cisco proprietary voice encapsulation for VoFR with data is carried on CID 4 and call-control on CID 5.
cid	(Optional) Specifies the subchannel to be used for call-control signaling. Valid range is from 4 to 255. The default is 5. If **call-control** is specified and a CID is not entered, the default CID is used.

Command Default

Disabled

Command Modes

Frame relay DLCI configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series routers and Cisco MC3810.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.

Release	Modification
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 Cisco IOS Release 12.1(2)T.

Usage Guidelines

The table below 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 1: 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]
Cisco-trunk permanent call (private-line) to other routers supporting VoFR	**vofr** **data** *cid* **call-control** *cid*
FRF.11 trunk call (private-line) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** *cid*][2]

1 The recommended form of this command to use is vofr data 4 call-control 5 .
2 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.

Examples

The following example, beginning in global configuration mode, shows how to enable VoFR on serial interface 1/1, DLCI 100. 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
```

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.

Command Default

Disabled

Command Modes

Service profile configuration mode

Command History

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

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.

voicecap configure

To apply a voicecap on NextPort platforms, use the **voicecap configure**command in voice-port configuration mode. To remove a voicecap, use the **no** form of this command.

voicecap configure *name*
no voicecap configure *name*

Syntax Description

name	Designates which voicecaps to use on this voice port.

Command Default

No default values or behavior

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The character value for the *name* argument must be identical to the value entered when you created the voicecap using the **voicecap entry** command.

Examples

The following example configures a voicecap with the name qualityERL:

```
Router> enable
Router# configure terminal
Router(config)# voicecap entry qualityERL v270=120
Router(config)# voice-port 3/0:D
Router(config-voiceport)# voicecap configure qualityERL
```

Related Commands

Command	Description
voicecap entry	Creates a voicecap on NextPort platforms.

voicecap entry

To create a voicecap, use the **voicecap entry** command in global configuration mode. To disable a voicecap, use the **no** form of this command.

voicecap entry [*name string*]
no voicecap entry [*name string*]

Syntax Description

name string	(Optional) A word and a string of characters that uniquely identify a voicecap. • The *name* argument specifies a unique identifier for a voicecap. • The *string* argument specifies one or more voicecap register entries, similar to a modemcap. Each entry is of the form **v***index* =*value* , where *index* refers to a specific V register, and *value* designates the value for that V register.

Command Default

No voice caps can be applied to configure firmware.

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.
12.3(11)T	This command was integrated into Cisco IOS Release 12.3(11)T.
12.4(4)XC	This command was modified to include GSMAMR-NB codec capability.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command configures firmware through voicecap strings. This command allows you to assign values to specific registers. Voicecaps are applied to specific voice ports at system startup.

The voicecap values can be entered in a DSP-recognizable format called *raw format* . They can also be entered in *standard format* , which allows you to use commonly accessible values, such as decibels.

Starting with Cisco IOS Release 12.4(4)XC, this command can be used to configure GSMAMR-NB codecs on Cisco AS5350XM and Cisco AS5400XM platforms. The register values for GSMAMR-NB are shown in the table below.

Table 2: GSMAMR-NB Register Values

V-Reg #	Default	Description	Register Values and Additional Notes
0	0	Sets how Adaptive Multi-Rate (AMR) responds to an incoming codec mode request (CMR) that is not a member of the mode set.	0 = Drop the packet with the bad CMR. 1 = Ignore the CMR (do not change rates) but process the rest of the packet data normally. 2 = Change the rate to the highest rate in the mode set lower than the rate requested by the CMR.

V-Reg #	Default	Description	Register Values and Additional Notes
1	0	Sets how AMR handles packets with a frame type (AMR rate) that is not a member of the mode set.	0 = Drop the packet with the bad frame-type. 1 = Attempt to decode the packet.

Examples

The following example creates a voicecap string for a GSMAMR-NB codec named gsmamrnb-ctrl with V register 0 set to 1:

```
Router> enable
Router# configure terminal

Router(config)# voicecap entry gsmamrnb-ctrl v0=1
```

Related Commands

Command	Description
voicecap configure	Applies a voicecap to the specified voice ports.

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
value	Minimum interval, in seconds, with a range of 1 to 3600 seconds and a default of 10. This value cannot be set higher than the time configured for the capacity update interval.

Command Default

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.

Command	Description
voice call capacity stw	Set the value for STW.

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.

Command Default

The capacity reporting function is turned off.

Command Modes

Global configuration.

Command History

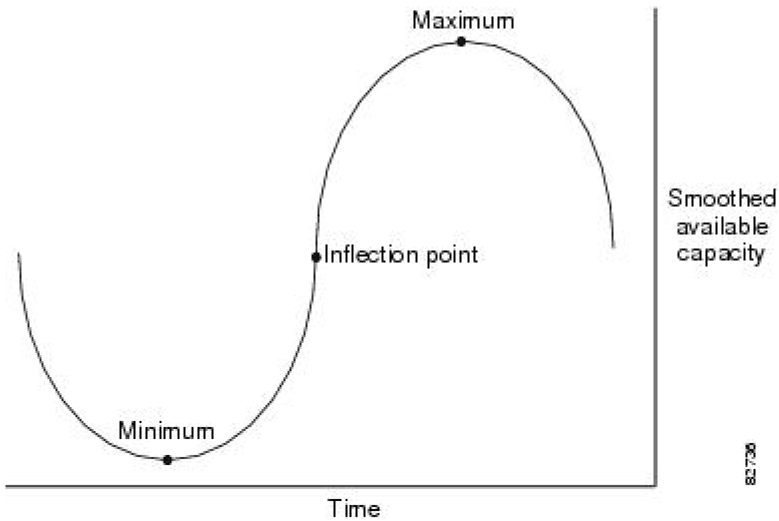
Release	Modification
12.3(1)	This command was introduced.

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 the figure below.

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 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	Transitions time can be from 0 to 60 seconds with a default of 10.

Command Default

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.

Command	Description
voice call capacity mir	Set the value for MIR.

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	Value from 10 to 3600 seconds.

Command Default

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.

Command Default

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 of the CSR data points, 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	Value from 10 to 50 data points. Default is 30 data points.

Command Default

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 the CSR.
voice call csr reporting interval	Sets the reporting interval for the 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 CSR recording 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	Value from 10 to 1000 minutes with a default of 60.

Command Default

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	Value from 10 to 10000 seconds with a default of 25.

Command Default

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 disable the **short-header** setting and return tothe **full-guid** setting, 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 the no version of this command is input 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.

Command Default

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 platforms: Cisco 2600 series, Cisco 3620, Cisco 3640, Cisco 3660 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, Cisco AS5850, and Cisco MC3810.
12.2(15)T	The header-only keyword was replaced by the short-header keyword.

Usage Guidelines

Despite its nontraditional syntax (trailing rather than preceding "debug"), this is a normal **debug** command.

You can control the contents of the standardized header. 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 is 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 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_from_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_canceller_control:
```

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

The table below describes significant fields shown in the display.

Table 3: voice call debug full-guid Field Descriptions

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

The following is sample output 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 "//-1/" output 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.

Command	Description
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 disc-pi-off

To enable the gateway to treat a disconnect message with progress indicator (PI) like a standard disconnect without a PI, use the **voice call disc-pi-off**command in global configuration mode. To reset to the default, use the **no** form of this command.

voice call disc-pi-off
no voice call disc-pi-off

Syntax Description

This command has no keywords or arguments.

Command Default

Gateway disconnects incoming call leg when it receives a disconnect message with PI.

Command Modes

Global configuration

Command History

Release	Modification
12.3(5)	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.

Usage Guidelines

Use this command if the gateway is connected to a switch that sends a release immediately after it receives a Disconnect with PI. To properly handle the call, the switch should open a backward voice path and keep the call active. Otherwise the rotary dial peer feature does not work because the incoming call leg is disconnected. Using this command enables the gateway to handle a disconnect with PI like a regular disconnect message so that you can use the rotary dial peer feature.

Examples

The following example enables the gateway to properly handle a disconnect with PI:

```
voice call disc-pi-off
```

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 convert-discpi-to-prog	Converts a disconnect message with a PI to a progress message.

voice call rate monitor

To enable voice call rate monitoring, use the **voice call rate monitor** command in voice service configuration mode. To disable voice call monitoring, use the **no** form of this command.

voice call rate monitor
no voice call rate monitor

Syntax Description

This command has no arguments or keywords.

Command Default

Voice call monitoring is disabled.

Command Modes

Voice service configuration (conf-voi-serv)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

You can use the **voice call rate monitor** command to enable the call monitoring functionality for a duration of 60 seconds.

Examples

The following example shows how to enable voice call rate monitoring on a Cisco Unified Border Element (Cisco UBE):

```
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# voice call rate monitor
```

Related Commands

Command	Description
show voice call rate	Displays the voice call rate information.

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.

Command Default

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 Cisco IOS Release 12.1(5.3)T.
12.2(1)	This command was integrated into Cisco IOS Release 12.2.

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 trap deviation

To configure the percentage deviation for voice call trap parameters, use the **voice call trap deviation** command in global configuration mode. To disable the configured percentage deviation, use the **no** form of this command.

voice call trap deviation *percent* [**vad**]
no voice call trap deviation *percent* [**vad**]

Syntax Description

percent	The percentage deviation for trapping calls. The range of acceptable values is 1 to 100. The default is 49.
vad	(Optional) Specifies the deviation for calls with voice activity detection (VAD) turned on.

Command Default

This command is enabled by default, and the deviation for trapping calls is set to 49 percent.

Command Modes

Global configuration (config)

Command History

Release	Modification
12.4(12)	This command was introduced in a release earlier than Cisco IOS Release 12.4(12).
15.0(1)M	The **no** form of this command was modified.

Usage Guidelines

Prior to Release 15.0(1)M, if a non-default *percent* value was configured, it could be disabled by entering the **no voice call trap deviation** *percent*command, even if the *percent* value was not the configured value. For example, if the **voice call trap deviation 30** command was configured, the **no voice call trap deviation 40**command disabled the initial command.

Beginning in Release 15.0(1)M, the *percent* value in the **no** form of the command must match the configured non-default value. For example, if the **voice call trap deviation 30** command is configured, the only way to disable it is to enter the **no voice call trap deviation 30**command. If the **no voice call trap deviation 40** command is entered, the command-line interface displays this message: "Please enter correct deviation."

Examples

The following example shows how to set the deviation value for trapping calls to 30 percent:

```
Router(config)# voice call trap deviation 30 vad
```

voice call trigger hwm

To set the value for 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	Value can be 50 to 100 percent with a default of 80. If set to 100, this trigger will be turned off.

Command Default

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

Command	Description
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	Value can be 0 to 30 percent with a default of 10. If set to 0, this trigger will be turned off.

Command Default

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

Command	Description
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 value for percentage change, low water mark and high water mark 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	If percent-change is selected, value can be 0 to 100 percent with a default of 30. If set to 0, this trigger will be turned off. If lwm is selected, value can be 0 to 30 percent with a default of 10. If set to 0, this trigger will be turned off. If hwm is select, value can be 50 to 100 percent with a default of 80. If set to 100, this trigger will be turned off.

Command Default

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.

Command	Description
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 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-card

To enter voice-card configuration mode and configure a voice card, use the **voice-card** command in global configuration mode. There is no **no** form of this command.

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 in the router chassis.
	  • 1 is the network module slot in the router chassis.
	• Cisco 3600 series:
	  • A value from 1 to 6 identifies a network module slot in the router chassis.
	• Cisco 3660:
	  • 7 is AIM slot 0 in the router chassis.
	  • 8 is AIM slot 1.

Command Default

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 Cisco IOS Release 12.0(7)T.
12.0(7)XK	This command was implemented on the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(2)XB	Values for the *slot* argument were updated to include AIMs.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(13)T	This command was supported in Cisco IOS Release 12.2(13)T and 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 Cisco IOS Release 12.2(15)T.

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.

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.

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:

```
voice-card 1
```

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 cause-code

To set the internal Q850 cause code mapping for voice and to enter voice cause configuration mode, use the **voice cause-code**command in global configuration mode. To disable the internal Q850 cause code mapping for voice, use the **no** form of this command.

voice cause-code
no voice cause-code

Syntax Description

This command has no arguments or keywords.

Command Default

Internal Q850 cause code mapping for voice is disabled.

Command Modes

Global configuration (config)

Command History

Release	Modification
15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Examples

The following example shows how to set the cause code mapping for voice:

```
Router> enable
Router# configure terminal
Router(config)# voice cause-code
```

Related Commands

Command	Description
voice class codec	Assigns an identification tag number for a codec voice class.

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	A number used to identify voice class AAA. The range is from 1 to 10000. There is no default value.

Command Default

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
accounting suppress	Disables accounting that is automatically generated by the service provider module for a specific dial peer.
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.

Command	Description
method	Specifies an accounting method for calls coming into the defined dial peer.
voice-class aaa	Applies properties defined in the voice class to a specific dial peer.

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 identification number assigned to one voice class. Range is 1 to 10000.

Command Default

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 series, Cisco 3600 series, 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.

Command	Description
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 called number

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

voice class called number {**inbound** | **outbound** | **pool**} *tag*
no voice class called number

Syntax Description

inbound	Inbound voice class called number.
outbound	Outbound voice class called number.
pool	Voice class called number pool.
tag	Digits that identify a specific inbound or outbound voice class called number or voice class called number pool.

Command Default

No voice class called number is configured.

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use this command to define one or more static voice class called numbers for inbound and outbound POTS dial peers or a dynamic voice class called number pool. The indexes for a voice class called number are defined with the **index** (voice class) command.

Note Enter the **voice class called number** command in global configuration mode without hyphens. Enter the **voice-class called-number** command in dial-peer configuration mode with hyphens.

Examples

The following example shows configuration for an outbound voice class called number:

```
voice class called number outbound 30
 index 1 5550100
 index 2 5550101
 index 3 5550102
 index 4 5550103
```

The following example shows configuration for a voice class called number pool:

```
voice class called number pool 1
 index 1 5550100 - 5550199
```

Related Commands

Command	Description
show voice class called-number	Displays a specific voice class called number.
voice-class called-number (dial-peer)	Assigns a previously defined voice class called number to an inbound or outbound POTS dial peer.

voice class cause-code

To configure cause code list parameters for a voice class and to enter cause code configuration mode, use the **voice class cause-code**command in global configuration mode. To disable the cause code list parameters configuration for a voice class, use the **no** form of this command.

voice class cause-code *number*
no voice class cause-code *number*

Syntax Description

number	Numeric tag that specifies the voice class cause code. The range is from 1 to 64.

Command Default

The cause code list parameters are not defined.

Command Modes

Global configuration (config)

Command History

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

Examples

The following example shows how to configure cause code list parameters for voice class 5:

```
Router> enable
Router# configure terminal
Router(config)# voice class cause-code 5
```

Related Commands

Command	Description
voice class codec	Assigns an identification tag number for a codec voice class.

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 number that you assign to the voice class. Range is 1–10000. There is no default.

Command Default

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 Cisco IOS Release 12.1(2)T.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
Cisco IOS XE Dublin 17.10.1a	Introduced support for the following YANG models in **codec preference** configuration: • **g729br8 [bytes <10-244>]** • **gsmamr-nb [encap | modes | packetization-period]** Introduced support for the following YANG model in **video codec** configuration: • **video codec [h261 | mpeg4]**

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 VoIP dial peer.

Note

- The **voice class codec**command in global configuration mode is entered without a hyphen. The **voice-class codec**command in dial-peer configuration mode is entered with a hyphen.
- **transparent** is not available under voice class codec in YANG. However, you can configure **codec transparent** directly under dial-peer.

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
codec preference 13 gsmamr-nb
```

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 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.

Command Default

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 platforms.
12.2(2)T	This command was implemented on Cisco 1750 access routers and integrated into Cisco IOS Release 12.2(2)T.

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 dscp-profile

To configure the differentiated services code point (DSCP) profile, use the **voice class dscp-profile** command in global configuration mode. To disable the configuration, use the **no** form of this command.

voice class dscp-profile *tag*
no voice class dscp-profile *tag*

Syntax Description

tag	Voice class DSCP tag. The range is from 1 to 10000.

Command Default

A DSCP profile is not configured.

Command Modes

Global configuration (config)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

You can use the **voice class dscp-profile** command to configure the DSCP profile and then configure DSCP policing and enter voice class configuration mode.

Examples

The following example shows how to configure a DSCP profile and enter voice class configuration mode:

```
Router> enable
Router# configure terminal
Router(config)# voice class dscp-profile 1
Router(config-class)# end
```

Related Commands

Command	Description
dscp media	Specifies the RPH to DSCP mapping.

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 identification number assigned to one voice class. Range is from 1 to 10000.

Command Default

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 series, and the 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 tag identification number assigned to a voice class. Range is from 1 to 10000.

Command Default

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 Cisco 1750 routers and integrated into Cisco IOS Release 12.2(2)T.

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 e164-pattern-map

To create an E.164 pattern map that specifies multiple destination E.164 patterns in a dial peer, use the **voice class e164-pattern map** command in global configuration mode. To remove an E.164 pattern map from a dial peer, use the **no** form of this command.

voice class e164-pattern-map *tag*
no voice class e164-pattern-map

Syntax Description

tag	A number assigned to a voice class E.164 pattern map. The range is from 1 to 10000.

Command Modes

Global configuration (config)

Command History

Release	Modification
15.2(4)M	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Examples

The following example shows how to create an E.164 pattern map that specifies multiple destination E.164 patterns in a dial peer:

```
Device(config)# voice class e164-pattern-map 2543
```

Related Commands

Command	Description
show voice class e164-pattern-map	Displays the configuration of E.164 pattern maps.
voice class e164-pattern-map load	Loads a destination E.164 pattern map that is specified by a text file on a dial peer.

voice-class dpg

To create a dial-peer group for grouping multiple outbound dial peers, use the **voice class dpg** command in global configuration mode.

voice class dpg *dial-peer-group-id*

Syntax Description

dial-peer-group-id	Assigns a tag for a particular dial-peer group. The range is 1-10000.

Command Default

Disabled by default.

Command Modes

Global configuration voice class (config).

Command History

Release	Modification
Cisco IOS 15.4(1)T Cisco IOS XE 3.11S	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

You can group up to 20 outbound (H.323, SIP or POTS) dial peers into a dial-peer group and configure this dial-peer group as the destination of an inbound dial peer. Once an incoming call is matched by an inbound dial peer with an active destination dial-peer group, dial peers from this group are used to route the incoming call. No other outbound dial-peer provisioning to select outbound dial peers is used.

A preference can be defined for each dial peer in a dial-peer group. This preference is used to decide the order of selection of dial peers from the group for the setup of an outgoing call.

You can also specify various dial-peer hunt mechanisms using the existing dial-peer hunt command. For more information, refer to Configure Outbound Dial-Peer Group as an Inbound Dial-Peer Destination.

Examples

```
Router(config)#voice class dpg ?
  <1-10000>  Voice class dialpeer group tag

Router(config)#voice class dpg 1
Router(config-class)#dial-pee
Router(config-class)#dial-peer ?
  <1-1073741823>  Voice dial-peer tag

Router(config-class)#dial-peer 1 ?
  preference  Preference order of this dialpeer in a group
  <cr>        <cr>

Router(config-class)#dial-peer 1 pre
Router(config-class)#dial-peer 1 preference ?
  <0-10>  Preference order

Router(config-class)#dial-peer 1 preference 9
Router(config-class)#
```

Related Commands

Command	Description
dial-peer voice	To define a dial peer.
destination -pattern	To configure a destination pattern.

voice class e164-pattern-map load

To load a destination E.164 pattern map that is specified by a text file on a dial peer, use the **voice class e164-pattern-map load** command in privileged EXEC mode.

voice class e164-pattern-map load *tag*

Syntax Description

tag	A number that is assigned to the destination E.164 pattern map. The range is from 1 to 10000.

Command Default

No default behavior or values.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
15.2(4)M	This command was introduced.

Usage Guidelines

After creating an E.164 pattern map, you can add destination E.164 pattern entries to the E.164 pattern map and store all the information on the voice gateway or create the E.164 pattern entries in a text file and store the file on the internally or externally supported file system.

Examples

The following example shows how to reload a particular destination E.164 pattern map on a dial peer:

```
Device# voice class e164-pattern-map load 2543
```

Related Commands

Command	Description
show voice class e164-pattern-map	Displays the configuration of E.164 pattern maps.
voice class e164-pattern-map	Creates an E.164 pattern map to specify multiple destination E.164 patterns in a dial peer.

voice class e164-translation

To translate the phone number of the call source into E.164 format, as per the translation rules, use the **voice class e164-translation** command in global configuration mode.

voice class e164-translation *tag*

Syntax Description

tag	The range is from 1 to 10000.

Command Modes

Global configuration (config)

Command History

Release	Modification
IOS XE Fuji Release 16.8.1	This command was introduced.

Example

The following example translates the input call number with tag 1 into E.164 format.

```
Router(config)# voice class e164-translation 1
Router(config-class)#url ftp://test:test123@8.0.0.200/test_e164.cfg
Router(config-class)#^Z
```

Related Commands

voice class e164-pattern-map	Creates an E.164 pattern map to specify multiple destination E.164 patterns in dial peer.
voice class e164-pattern-map load	Loads a destination E.164 pattern map that is specified by a text file on a dial peer.

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 number to identify the voice class. Range is from 1 to 10000. There is no default value.

Command Default

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 series, 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 demonstrates how a voice class is created and applied to an individual dial peer. Voice class 4 contains a command to disable the capability to detect Cisco CallManager systems in the network (this command is used by Cisco CallManager Express 3.1 and later versions). The example then uses the **voice-class h323**command to apply voice class 4 to dial peer 36.

```
Router(config)# voice class h323 4
Router(config-class)# no telephony-service ccm-compatible
Router(config-class)# exit
Router(config)# dial-peer voice 36 voip
Router(config-dial-peer)# destination-pattern 555....
Router(config-dial-peer)# session target ipv4:10.5.6.7

Router(config-dial-peer)# voice-class h323 4
```

Related Commands

Command	Description
voice-class h323	Assigns an H.323 voice class to a VoIP dial peer.

voice class media

To configure the media control parameters for voice, use the **voice class media**command in global configuration mode. To disable the media control parameters for voice, use the **no** form of this command.

voice class media *number*
no voice class media *number*

Syntax Description

number	Numeric tag that specifies the voice class media. The range is from 1 to 10000.

Command Default

The media control parameters for voice are not configured.

Command Modes

Global configuration (config)

Command History

Release	Modification
15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Examples

The following example shows how to configure media control parameters for voice:

```
Router> enable
Router# configure terminal
Router(config)# voice class media 5
```

Related Commands

Command	Description
voice class codec	Assigns an identification tag number for a codec 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 number that you assign to the voice class. Range is from 1 to 10000.

Command Default

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 Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was implemented on 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.

Command	Description
voice-class permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a network dial peer.

voice class resource-group

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

voice class resource-group *tag*
no voice class resource-group *tag*

Syntax Description

tag	Unique tag to identify the resource. The range is from 1 to 5.

Command Default

No resource groups are created.

Command Modes

Global configuration (config)

Command History

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

Usage Guidelines

Use the **voice class resource-group** command to configure parameters along with the threshold values to be monitored for resource groups. When you use the **voice class resource-group** command, the router enters voice-class configuration mode. You can then group the resources to be monitored and configure parameters such as .

Examples

The following example shows how to enter voice-class configuration mode and assign identification tag number 5 for a resource group:

```
Router> enable
Router# configure terminal
Router(config)# voice class resource-group 5
```

Related Commands

Command	Description
debug rai	Enables debugging for Resource Allocation Indication (RAI).
periodic-report interval	Configures periodic reporting parameters for gateway resource entities.
rai target	Configures the SIP RAI mechanism.
resource (voice)	Configures parameters for monitoring resources.
show voice class resource-group	Displays the resource group configuration information for a specific resource group or all resource groups.

voice class route-string

To assign a unique identifier tag to a route string, use the **voice class route-string** command in global configuration mode. To remove the route string, use the **no** form for this command.

voice class route-string *tag*
no voice class route-string *tag*

Syntax Description

tag	Unique tag to identify the route string. The range is from 1 to 10000.

Command Default

An identifier tag for the route string is not configured.

Command Modes

Global configuration (config)

Command History

Release	Modification
15.3(3)M	This command was introduced.
Cisco IOS XE Release 3.10S	This command was integrated into Cisco IOS XE Release 3.10S.

Usage Guidelines

Use the **voice class route-string** command to assign a unique identification tag to the route string. You can use this command to enter voice class configuration mode to configure a route string pattern.

Example

The following example shows how to assign identification tag 2 for a route string:

```
Device> enable
Device# configure terminal
Device(config)# voice class route-string 2
Device(config-class)# pattern london.uk.eu
```

voice class server-group

To enter voice-class configuration mode and configure server groups (groups of IPv4 and IPv6 addresses) which can be referenced from an outbound SIP dial peer, use the **voice class server-group** command in global configuration mode. To delete a server group, use the **no** form of this command.

voice class server-group *server-group-id*
no voice class server-group *server-group-id*

server-group-id	Unique server group ID to identify the server group. You can configure up to five servers per server group.

Command Default

No server groups are created.

Command Modes

Global configuration (config)

Release	Modification
Cisco IOS XE Release 3.11S 15.4(1)T	The following commands were introduced or modified: **voice class server-group**, **description**, **ipv4 port preference**, **ipv6 port preference**, **hunt-scheme**, **show voice class server-group**, **shutdown (Server Group)**.
Cisco IOS XE Bengaluru 17.4.1a	The following command is introduced under **voice class server-group**. **huntstop** *rule-tag* **resp-code** *from_resp_code* to *to_resp_code*.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use the **voice class server-group** command to group IPv4 and IPv6 addresses of servers and configure it as in an outbound SIP dial-peer. When you use the **voice class server-group** command, the router enters voice-class configuration mode. You can then group the servers and associate them with a SIP outbound dial peer.

The following example shows how to enter voice-class configuration mode and assign server group id for a server group:

```
Router> enable
Router# configure terminal
Router(config)# voice class server-group 2
```

After configuring a voice class server-group, you can configure a server IP address along with an optional port number and preference, as part of this server group along with an optional port number and preference order. You can also configure description, hunt-scheme, and huntstop. You can use the shutdown command to make the server group inactive.

```
Device(config)# voice class server-group 2
Device(config-class)# ipv4 10.1.1.1 preference 1
Device(config-class)# ipv4 10.1.1.2 preference 2
Device(config-class)# ipv4 10.1.1.3 preference 3
Device(config-class)# description It has 3 entries
Device(config-class)# hunt-scheme round-robin
```

```
Device(config-class)# huntstop 1 resp-code 400 to 599
Device(config-class)# exit
```

Related Commands

Command	Description
description	Provides a description for the server group.
hunt-scheme	Defines a hunt method for the order of selection of target server IP addresses (from the IP addresses configured for this server group) for the setting up of outgoing calls.
shutdown (Server Group)	To make the server group inactive.
show voice class server-group	Displays the configurations for all configured server groups or a specified server group.

voice class sip-copylist

To configure a list of entities to be sent to the peer call leg, use the **voice class sip-copylist** command in global configuration mode. To disable the configuration, use the **no** form of this command.

voice class sip-copylist *tag*
no voice class sip-copylist *tag*

Syntax Description

tag	Voice class Session Initiation Protocol (SIP) copylist tag. The range is from 1 to 10000.

Command Default

No header is sent to the peer call leg.

Command Modes

Global configuration (config).

Voice class tenant configuration (config-class).

Command History

Release	Modification
15.1(3)T	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use the **voice class sip-copylist** command to configure Cisco Unified Border Element (UBE) to pass an unsupported parameter present in a mandatory header from one call leg to another of Cisco UBE. You can copy the inbound message headers into variables and pass the headers to the outbound call leg.

Examples

The following example shows how to configure a SIP list to be sent to the peer call leg:

```
Router(config)# voice class sip-copylist 5
```

Related Commands

Command	Description
sip-header	Specifies the SIP header to be sent to the peer call leg.

voice class sip-hdr-passthrulist

To configure a list of headers to pass-through, use the **voice class sip-hdr-passthrulist** command in global configuration mode. To remove the header pass-through, use the **no** form of this command.

voice class sip-hdr-passthrulist *tag*
no voice class sip-hdr-passthrulist *tag*

Syntax Description

tag	Unique tag to identify the header. The range is from 1 to 1000.

Command Default

None

Command Modes

Global configuration (config)

Voice class tenant configuration (config-class)

Command History

Release	Modification
15.3(3)M	This command was introduced.
Cisco IOS XE Release 3.10S	This command was integrated into Cisco IOS XE Release 3.10S.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use the **voice class sip-hdr-passthrulist** command to configure a list of headers to be passed-through the route string. You can use this command to enter the voice class configuration mode to configure route-string header pass-through.

Example

The following example shows how to configure header pass-through with the unique identification tag 2:

```
Device> enable
Device# configure terminal
Device(config)# voice class sip-hdr-passthrulist 2
Device(config-class)# passthru-hdr x-cisco-dest-route-string
Device(config-class)# passthru-hdr Supported
Device(config-class)# passthru-hdr Subject
```

voice class sip-profiles

To configure Session Initiation Protocol (SIP) profiles for a voice class, use the **voice class sip-profiles**command in global configuration mode. To disable the SIP profiles for a voice class, use the **no** form of this command.

voice class sip-profiles *number*
no voice class sip-profiles *number*

Syntax Description

number	Numeric tag that specifies the voice class SIP profile. The range is from 1 to 10000.

Command Default

SIP profiles for a voice class are not configured.

Command Modes

Global configuration (config).

Voice class tenant configuration (config-class).

Command History

Release	Modification
15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Note The rule option **[before]** is not available in sip-profile YANG configuration.

voice class sip-profile <tag>

rule [before]

Examples

The following example shows how to specify SIP profile 2 for a voice class:

```
Router> enable
Router# configure terminal
Router(config)# voice class sip-profiles 2
```

Related Commands

Command	Description
voice class codec	Assigns an identification tag number for a codec voice class.

voice class srtp-crypto

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

voice class srtp-crypto *tag*
no voice class srtp-crypto *tag*

Syntax Description

tag	Unique number that you assign to the srtp-crypto voice class. Range is 1–10000. There is no default.

Command Default

No default behavior or values.

Command Modes

Global configuration (config).

Command History

Release	Modification
Cisco IOS XE Everest 16.5.1b	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
Cisco IOS XE Dublin 17.10.1a	The YANG model for this command can now be configured under **voice register pool**.

Usage Guidelines

This command only creates the voice class for srtp-crypto preference selection and assigns an identification tag. Use the **crypto** command under voice class srtp-crypto submode to select the ordered list of preferred cipher-suites.

Deleting srtp-crypto voice class using **no voice class srtp-crypto** *tag* command removes the srtp-crypto tag (same tag) if configured in global, tenant, and dial-peer configuration mode.

Example

```
Device> enable
Device# configure terminal
Device(config)# voice class srtp-crypto 100
```

Related Commands

Command	Description
srtp-crypto	Assigns a previously configured crypto-suite selection preference list globally or to a voice class tenant.
crypto	Specifies the preference for an SRTP cipher-suite that will be offered by Cisco Unified Border Element (CUBE) in the SDP in offer and answer.
show sip-ua calls	Displays active user agent client (UAC) and user agent server (UAS) information on Session Initiation Protocol (SIP) calls.

Command	Description
show sip-ua srtp	Displays Session Initiation Protocol (SIP) user-agent (UA) Secure Real-time Transport Protocol (SRTP) information.

voice class tenant

To enter voice class tenant configuration mode and to allow tenants to configure their own global configurations for a specific voice class, use the **voice class tenant** command in global configuration mode. To disable the tenant configurations for a voice class, use the **no** form of this command.

voice class tenant *tag*
no voice class tenant *tag*

Syntax Description

tag	A number used to identify voice class tenant. The range is from 1 to 10000. There is no default value.

Command Default

No default behavior or values.

Command Modes

Global configuration (config)

Command History

Release	Modification
15.6(2)T and IOS XE Denali 16.3.1	This command was introduced.
Cisco IOS XE Bengaluru 17.4.1a	Introduced support for YANG models.

Usage Guidelines

The **voice class tenant** command sets up a voice service class that allows tenants to configure their own sip-specific configurations.

Examples

The following example shows how to configure tenants for a voice class:

```
Device(config)# voice class tenant 1
Device (config-class)# ?
aaa – sip-ua AAA related configuration
anat – Allow alternative network address types IPV4 and IPV6
asserted-id – Configure SIP-UA privacy identity settings
……
……
Video – video related function
Warn-header – SIP related config for SIP. SIP warning-header global config
Device (config-voi-tenant)# end
```

voice class tls-profile

To enable voice class configuration mode, and assign an identification tag for a TLS profile, use the command **voice class tls-profile** in global cofiguration mode. To remove a tls-profile, use the **no** form of this command.

voice class tls-profile *tag*

no voice class tls-profile *tag*

Syntax Description

tag	A number used to identify voice class TLS profile. The range is 1-10000. There is no default value.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
Cisco IOS XE Amsterdam 17.3.1a	This command was introduced.

Usage Guidelines

The command **voice class tls-profile** enables voice class configuration mode on the router and provides you sub-options to configure commands required for a TLS session. This command allows you to configure under voice class, the options that can be configured at the global level via sip-ua.

The *tag* associates all the voice class configurations that are made through the command **voice class tls-profile** *tag* to the command **crypto signaling**. Following is the **crypto signaling** command with **tls-profile** *tag*:

crypto signaling {**remote-addr** *ip address subnet mask* | **default**} **tls-profile** *tag*

For more information on the updates to the command **crypto signaling**, see crypto signaling.

Examples

The following example configures the **voice class tls-profile** with *tag* '2' and enables voice class cofiguration mode:

```
Router(config)#voice class tls-profile 2
Router(config-class)#
```

The following section provides details of the sub-commands that can be configured under the command **voice class tls-profile** *tag*.

The following example configures CUBE to use the **trustpoint** *trustpoint-name* keyword and argument when it establishes or accepts the TLS connection with a remote device:

```
Router(config-class)#trustpoint CUBETP
```

The following example configures client verification trustpoint:

```
Router(config-class)#client-vtp TPname
```

The following example indicates the description for the TLS profile group:

```
Router(config-class)#description tlsgroupname
```

The following example configures the specific size of elliptic curves to be used for a TLS session:

```
Router(config-class)#cipher ecdsa-cipher curve-size 384
```

The following example configures CUBE to perform server identity validation through Common Name (CN) and Subject Alternate Name (SAN) fields in the server certificate:

```
Router(config-class)#cn-san-validate server
```

The following example enables Server Name Indication (SNI) required during the initial TLS handshake process:

```
Router(config-class)#sni send
```

Related Commands

Command	Description
trustpoint	Creates a trustpoint to store the devices certificate that is generated as part of the enrollment process using Cisco IOS public-key infrastructure (PKI) commands.
description	Provides a description for the TLS profile group.
client-vtp	Assigns a client verification trustpoint.
cipher	Configures cipher setting.
cn-san	Enables server identity validation through Common Name (CN) and Subject Alternate Name (SAN) fields in the server certificate during client-side SIP /TLS connections
sni send	Enables TLS Server Name Indication (SNI) during the initial TLS handshake process.
crypto signaling	Identifies the trustpoint or the **tls-profile** *tag* that is used during the TLS handshake process.

voice class tls-cipher

To configure an ordered set of TLS cipher suites, use **voice class tls-cipher** command. To disable this command or revert to default, use the **no** form of this command.

voice class tls-cipher *tag*
no voice class tls-cipher *tag*

tag	Specifies the voice class tls-cipher tag.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Release	Modification
Cisco IOS XE Cupertino 17.8.1a	This command was introduced.

Usage Guidelines

The **voice class tls-cipher** command enables voice class configuration mode on the router, allowing you to configure an ordered list of TLS cipher suites:

Examples

```
Router(config)#voice class tls-cipher 123

Router(config-class)# cipher ?
    <1-10>  Set the preference order for the cipher-suite (1 = Highest)

Router(config-class)#cipher 1 ?

AES128_GCM_SHA256               supported in TLS 1.3
AES256_GCM_SHA384               supported in TLS 1.3
CHACHA20_POLY1305_SHA256        supported in TLS 1.3
DHE_RSA_AES128_GCM_SHA256       supported in TLS 1.2
DHE_RSA_AES256_GCM_SHA384       supported in TLS 1.2
DHE_RSA_WITH_AES_128_CBC_SHA    supported in TLS 1.2 & below
DHE_RSA_WITH_AES_256_CBC_SHA    supported in TLS 1.2 & below
ECDHE_ECDSA_AES128_GCM_SHA256   supported in TLS 1.2
ECDHE_ECDSA_AES256_GCM_SHA384   supported in TLS 1.2
ECDHE_RSA_AES128_GCM_SHA256     supported in TLS 1.2
ECDHE_RSA_AES256_GCM_SHA384     supported in TLS 1.2
RSA_WITH_AES_128_CBC_SHA        supported in TLS 1.2 & below
RSA_WITH_AES_256_CBC_SHA        supported in TLS 1.2 & below

Router(config-class)# cipher 1 ECDHE_RSA_AES256_GCM_SHA384 ?
<cr>   <cr>
Router(config-class)#
```

voice class tone-signal

To enter voice-class configuration mode and create a tone-signal voice class, use the **voice class tone-signal** command in global configuration mode. To delete a tone-signal voice class, use the **no** form of this command.

voice class tone-signal *tag*
no voice class tone-signal *tag*

Syntax Description

tag	Label that uniquely identifies the voice class. Can be up to 32 alphanumeric characters.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the **voice class tone-signal** command to define wakeup, frequency selection, and guard tones to be played out before and during the voice packets for a specific voice port. Use the **inject guard-tone**, **inject pause**, and **inject tone** commands to define the tone signaling in this class. You can configure up to ten tones in a tone-signal voice class.

To avoid voice loss at the receiving end of an LMR system, the maximum of the sum of the durations of the injected tones and pauses in the voice class should not exceed 1500 milliseconds. You must also use the **timing delay-voice tdm** command to configure a delay for the voice packet equal to the sum of the durations of all the injected tones and pauses.

Note that the hyphenation in this command differs from the hyphenation used in a similar command, **voice-class tone-signal**, which is used in voice-port configuration mode.

Examples

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

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

Related Commands

Command	Description
inject guard-tone	Plays out a guard tone with the voice packet.
inject pause	Specifies a pause between injected tones.

Command	Description
inject tone	Specifies a wakeup or frequency selection tone to be played out before the voice packet.
timing delay-voice tdm	Specifies the delay before a voice packet is played out.
voice-class tone-signal	Assigns a previously configured tone-signal voice class to a voice port.

voice class uri

To create or modify a voice class for matching dial peers to a Session Initiation Protocol (SIP) or telephone (TEL) uniform resource identifier (URI), use the **voice class uri**command in global configuration mode. To remove the voice class, use the **no** form of this command.

voice class uri *tag* {**sip** | **tel**}
no voice class uri *tag*

Syntax Description

tag	Label that uniquely identifies the voice class. Can be up to 32 alphanumeric characters.
sip	Voice class for SIP URIs.
tel	Voice class for TEL URIs.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

- This command takes you to voice URI class configuration mode, where you configure the match characteristics for a URI. The commands that you enter in this mode define the set of rules by which the URI in a call is matched to a dial peer.
- To reference this voice class for incoming calls, use the **incoming uri** command in the inbound dial peer. To reference this voice class for outgoing calls, use the **destination uri** command in the outbound dial peer.
- Using the **no voice class uri** command removes the voice class from any dial peer where it is configured with the **destination uri** or **incoming uri** commands.

Examples

The following example defines a voice class for SIP URIs:

```
voice class uri r100 sip
 user-id abc123
 host server1
 phone context 408
```

The following example defines a voice class for TEL URIs:

```
voice class uri r101 tel
 phone number ^408
 phone context 408
```

Related Commands

Command	Description
debug voice uri	Displays debugging messages related to URI voice classes.
destination uri	Specifies the voice class used to match the dial peer to the destination URI for an outgoing call.
host	Matches a call based on the host field in a SIP URI.
incoming uri	Specifies the voice class used to match a VoIP dial peer to the URI of an incoming call.
pattern	Matches a call based on the entire SIP or TEL URI.
phone context	Filters out URIs that do not contain a phone-context field that matches the configured pattern.
phone number	Matches a call based on the phone number field in a TEL URI.
show dialplan incall uri	Displays which dial peer is matched for a specific URI in an incoming call.
show dialplan uri	Displays which outbound dial peer is matched for a specific destination URI.
user-id	Matches a call based on the user-id field in the SIP URI.

voice class uri sip preference

To set the preference for selecting a voice class for Session Initiation Protocol (SIP) uniform resource identifiers (URIs), use the **voice class uri sip preference**command in global configuration mode. To reset to the default, use the **no** form of this command.

voice class uri sip preference {**user-id** | **host**}
no voice class uri sip preference

Syntax Description

user-id	User-id field is given preference.
host	Host field is given preference.

Command Default

Host field

Command Modes

Global configuration

Command History

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

Usage Guidelines

- Use this command to resolve ties when more than one voice class is matched for a SIP URI. The default is to match on the host field of the URI.
- This command applies globally to all URI voice classes for SIP.

Examples

The following example defines the preference as the user-id for a SIP voice class:

```
voice class uri sip preference user-id
```

Related Commands

Command	Description
debug voice uri	Displays debugging messages related to URI voice classes.
destination uri	Specifies the voice class used to match the dial peer to the destination URI for an outgoing call.
host	Matches a call based on the host field in a SIP URI.
incoming uri	Specifies the voice class used to match a VoIP dial peer to the URI of an incoming call.
user-id	Matches a call based on the user-id field in the SIP URI.
show dialplan incall uri	Displays which dial peer is matched for a specific URI in an incoming call.
show dialplan uri	Displays which outbound dial peer is matched for a specific destination URI.

Command	Description
voice class uri	Creates or modifies a voice class for matching dial peers to a SIP or TEL URI.

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	A number to identify the voice class. Range is from 1 to 10000. There is no default.

Command Default

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 this command.

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 this command.

```
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 called-number (dial peer)

To assign a previously defined voice class called number to an inbound or outbound POTS dial peer, use the **voice-class called-number**command in dial peer configuration mode. To remove a voice class called number from the dial peer, use the **no** form of this command.

voice-class called-number [{**inbound** | **outbound**}] *tag*
no voice-class called-number

Syntax Description

inbound	Assigns an inbound voice class called number to the dial peer.
outbound	Assigns an outbound voice class called number to the dial peer.
tag	Digits that identify a specific voice class called number.

Command Default

No voice class called number is configured on the dial peer.

Command Modes

Dial peer configuration

Command History

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

Usage Guidelines

Use this command to assign a previously defined voice class called number to a dial peer for a static H.320 secondary call dial plan. Use the **inbound** keyword for inbound POTS dial peers, and the **outbound** keyword for outbound POTS dial peers.

Note The **voice class called number** command in global configuration mode is entered without hyphens. The **voice-class called-number** command in dial peer configuration mode is entered with hyphens.

Examples

The following example shows configuration for an outbound voice class called number outbound on POTS dial peer 22:

```
dial-peer voice 22 pots
 voice-class called-number inbound 300
```

Related Commands

Command	Description
voice class called number	Defines a voice class called number or range of numbers for H.320 calls.
voice-class called-number-pool	Defines a pool of dynamic voice class called numbers for a voice port.

voice-class called-number-pool

To assign a previously defined voice class called number pool to a voice port, use the **voice-class called-number-pool** command in voice class configuration mode. To remove a voice class called number pool from the voice port, use the **no** form of this command.

voice-class called-number-pool *tag*
no voice-class called-number-pool

Syntax Description

tag	Digits that identify a specific voice class called number pool.

Command Default

No voice class called number pool is assigned to the voice port.

Command Modes

Voice class configuration

Command History

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

Usage Guidelines

Use this command to assign a voice class called number pool to a voice port for a dynamic H.320 secondary call dial plan.

Examples

The following example shows configuration for voice class called number pool 100 on voice port 1/0/0:

```
voice-port 1/0/0
 voice-class called-number-pool 100
```

Related Commands

Command	Description
voice class called number	Defines a voice class called number or range of numbers for H.320 calls.
voice-class called-number (dial peer)	Defines a called number or range of called numbers for a POTS dial peer.

voice-class codec (dial peer)

To assign a previously configured codec selection preference list (codec voice class) to a 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* [**offer-all**]
no voice-class codec

Syntax Description

tag	Unique number assigned to the voice class. The range is from 1 to 10000. • This tag number maps to the tag number created using the **voice class codec**command available in global configuration mode.
offer-all	(Optional) Adds all the configured codecs from the voice class codec to the outgoing offer from the Cisco Unified Border Element (Cisco UBE).

Command Default

Dial peers have no codec voice class assigned.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Release	Modification
12.0(2)XH	This command was introduced in Cisco IOS Release 12.0(2)XH and implemented on the Cisco AS5300 series routers.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T and implemented on the Cisco 2600 series and the Cisco 3600 series.
12.0(7)XK	This command was integrated into Cisco IOS Release 12.0(7)XK and implemented on the Cisco MC3810.
15.1(2)T	This command was modified. The **offer-all** keyword was added.
Cisco IOS XE Release 2.5	This command was integrated into Cisco IOS XE Release 2.5.
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

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:

```
Router# configure terminal

Router(config)# dial-peer voice 100 voip
Router(config-dial-peer)# voice-class codec 10 offer-all
```

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 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 number to identify the voice class. Range is from 1 to 10000.

Command Default

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 demonstrates how a voice class is created and applied to an individual dial peer. Voice class 4 contains a command to disable the capability to detect Cisco CallManager systems in the network (this command is used by Cisco CallManager Express 3.1 and later versions). The example then uses the **voice-class h323**command to apply voice class 4 to dial peer 36.

```
Router(config)# voice class h323 4
Router(config-class)# no telephony-service ccm-compatible
Router(config-class)# exit
Router(config)# dial-peer voice 36 voip
Router(config-dial-peer)# destination-pattern 555....
Router(config-dial-peer)# session target ipv4:10.5.6.7

Router(config-dial-peer)# voice-class h323 4
```

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 (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 number 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.

Command Default

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 Cisco MC3810.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was implemented on 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.

Command	Description
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 number 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.

Command Default

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 Cisco MC3810.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was implemented as a voice-port configuration command on Cisco 2600 series and Cisco 3600 series routers.

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 1/1/0:

```
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.

Command	Description
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.

voice-class sip anat

To enable Alternative Network Address Types (ANAT) on a Session Initiation Protocol (SIP) trunk, use the **voice-class sip anat** command in SIP configuration or dial peer configuration mode. To disable ANAT on SIP trunks, use the **no** form of this command.

voice-class sip anat [**system**]
no voice-class sip anat [**system**]

Syntax Description

system	(Optional) Configures ANAT globally.

Command Default

ANAT is enabled on SIP trunks.

Command Modes

SIP configuration (conf-serv-sip)
Dial peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

Both the Cisco IOS SIP gateway and Cisco Unified Border Element are required to support the Session Description Protocol (SDP) ANAT semantics. The **bind** command allows the use of ANAT semantics in outbound SDP. SDP ANAT semantics are intended to address scenarios that involve different network address families (for example, different IPv4 versions). Media lines grouped using ANAT semantics provide alternative network addresses of different families for a single logical media stream. The entity creating a session description with an ANAT group must be ready to receive or send media over any of the grouped "m" lines.

By default, ANAT is enabled on SIP trunks. However, if the SIP gateway is configured in IPv4-only mode or IPv6-only mode, the gateway will not use ANAT semantics in its SDP offer.

The **system** keyword configures ANAT on all network dial peers, including the local dial peer. Using the **voice-class sip anat** command without the **system** keyword enables ANAT only for the local dial peer.

Examples

The following example globally enables ANAT on a SIP trunk:

```
Router(config-serv-sip)# voice-class sip anat system
```

The following example enables ANAT on a specified dial peer:

```
Router(config-dial-peer)# voice-class sip anat
```

Related Commands

Command	Description
bind	Binds the source address for signaling and media packets to the IPv4 or IPv6 address of a specific interface.

voice pcm capture

To allocate the number of Pulse Code Modulation (PCM) capture buffers, to set up or change the destination URL for captured data, to enable PCM capture on-demand, and to change the PCM capture trigger string by the user, use the **voice pcm 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 number of buffers to 0, use the **no** form of this command.

voice pcm capture {**buffer** *number* | **destination** *url* | **on-demand-trigger** | **user-trigger-string** *start-string stop-string* **stream** *bitmap* **duration** *call-duration*}
no voice pcm capture {**buffer** *number* | **destination** *url* | **on-demand-trigger** | **user-trigger-string**}

Syntax Description

buffer *number*	Allocates the number of PCM capture buffers. The range is from 0 to 200000. The default is 0.
destination *url*	Specifies the destination URL for storing captured data.
on-demand-trigger	(Optional) Configures PCM capture user trigger on-demand.
user-trigger-string *start-string stop-string* **stream** *bitmap* **duration** *call-duration*	(Optional) Configures PCM user trigger string. • *start-string*—Start string up to 15 characters. • *stop-string*—Stop string up to 15 characters. • **stream**—Configures the PCM capture stream bitmap. • *bitmap*—PCM stream bitmap in hexadecimal. The range is from 1 to FFFFFFF. The default is 7. • **duration**—Configures the duration for PCM capture. • *call-duration*—Duration of call. The range is from 0 to 255. The default is 0.

Command Default

The default values are as follows:

- Number of buffers: 0
- Start string: 123
- Stop string: 456
- Stream: 7
- Call duration: 0

Command Modes

Global configuration (config)

Command History

Release	Modification
15.2(2)T	This command was introduced.

Usage Guidelines

If you want to change the number of an existing nonzero buffer, you must first reset it to 0 and then change it from 0 to the new number.

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 this 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 pcm 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.

Examples

The following example shows how to configure the number of PCM capture buffers:

```
Router> enable
Router# configure terminal
Router(config)# voice pcm capture buffer 200
```

The following example shows how to configure the destination URL for storing captured data:

```
Router> enable
Router# configure terminal
Router(config)# voice pcm capture destination tftp://10.0.1.10/acphan/
```

The following example shows how to configure user trigger PCM capture:

```
Router> enable
Router# configure terminal
Router(config)# voice pcm capture on-demand-trigger
```

The following example shows how to change the default user trigger PCM capture start and stop string, stream, and call duration:

```
Router> enable
Router# configure terminal
Router(config)# voice pcm capture #132 #543 stream ff duration 230
```

Related Commands

Command	Description
show voice pcm capture	Displays PCM capture status and statistics.

voice-class sip asserted-id

To enable support for the dial-peer-based asserted ID header in incoming Session Initiation Protocol (SIP) requests or response messages, and to send asserted ID privacy information in outgoing SIP requests or response messages, use the **voice-class sip asserted-id** command in dial-peer configuration mode. To disable the support for the asserted ID header, use the **no** form of this command.

voice-class sip asserted-id {**pai** | **ppi** | **system**}
no voice-class sip asserted-id

Syntax Description

pai	(Optional) Enables the P-Asserted-Identity (PAI) privacy header in incoming and outgoing SIP requests or response messages.
ppi	(Optional) Enables the P-Preferred-Identity (PPI) privacy header in incoming SIP requests and outgoing SIP requests or response messages.
system	(Optional) Uses global-level configuration settings to configure the dial peer.

Command Default

The privacy information is sent using the Remote-Party-ID (RPID) header or the FROM header.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Release	Modification
15.1(1)T	This command was introduced.
15.1(3)T	This command was modified. Support for incoming calls was added.

Usage Guidelines

If you choose the **pai** keyword or the **ppi** keyword for incoming messages, the gateway builds the PAI or the PPI header, respectively, into the common SIP stack, thereby sending the call data using the PAI or the PPI header. For outgoing messages, the privacy information is sent on the PAI or PPI header. The **pai** keyword or the **ppi** keyword has priority over the Remote-Party-ID (RPID) header, and removes the RPID/FROM header from the outbound message, even if the router is configured to use the RPID header at the global level.

Examples

The following example shows how to enable support for the PPI header:

```
Router> enable
Router# configure terminal
Router(config)# dial peer voice 1
Router(conf-voi-serv)# voice-class sip asserted-id ppi
```

Related Commands

Command	Description
asserted-id	Enables support for the asserted ID header in incoming and outgoing SIP requests or response messages at the global level.
calling-info pstn-to-sip	Specifies calling information treatment for PSTN-to-SIP calls.

Command	Description
privacy	Sets privacy in support of RFC 3323.

voice-class sip associate registered-number

To associate the preloaded route and outbound proxy details to the registered number in the dail peer configuration mode, use the **voice-class sip associate registered-number**command in dial peer configuration mode. To remove the association, use the **no** form of this command.

voice-class sip associate registered-number *number* [**system**]
no voice-class sip associate registered-number

Syntax Description

number	Registered number. The number must be between 4 and 32.
system	(Optional) Configures the association globally.

Command Default

The preloaded route and outbound proxy details are not associated by default.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

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

Usage Guidelines

The **voice-class sip associate registered-number**command takes precedence over the **associate registered-number**command in voice service VOIP SIP configuration mode. However, if the **voice-class sip associate registered-number** command is used with the **system** keyword, the gateway uses the settings configured globally by the **associate registered-number**command.

Examples

The following example shows how to associate a registered number on dial peer.

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2611 voip
Router(config-dial-peer)# voice-class sip associate registered-number 20
```

Related Commands

Command	Description
associate registered- number	Associates the preloaded route and outbound proxy details with the registered number in voice service VoIP SIP configuration mode.

voice-class sip asymmetric payload

To configure dynamic Session Initiation Protocol (SIP) asymmetric payload support on a dial peer, use the **voice-class sip asymmetric payload**command in dial peer configuration mode. To disable the configuration, use the **no** form of this command.

voice-class sip asymmetric payload {**dtmf** | **dynamic-codecs** | **full** | **system**}
no voice-class sip asymmetric payload

Syntax Description

dtmf	Provides asymmetric support only for dual-tone multi-frequency (DTMF) payloads.
dynamic-codecs	Provides asymmetric support only for dynamic codec payloads.
full	Provides asymmetric support both for DTMF and dynamic codec payloads.
system	(Optional) Specifies that the asymmetric payload uses the global value.

Command Default

Disabled (dynamic SIP asymmetric payload support is not enabled).

Command Modes

Dial peer (config-dial-peer)

Command History

Release	Modification
12.4(15)T	This command was introduced.
Cisco IOS XE Release 3.1S	This command was integrated into Cisco IOS Release IOS XE 3.1S

Usage Guidelines

For the Cisco UBE the SIP asymmetric payload-type is supported for audio/video codecs, DTMF, and NSE. Hence, **dtmf** and **dynamic-codecs** keywords are internally mapped to the **full** keyword to provide asymmetric payload-type support for audio/video codecs , DTMF, and NSE.

Examples

The following example shows how to configure dynamic SIP asymmetric payload support:

```
Router# configure terminal
Router(config)# dial-peer voice 77 voip
Router(config-dial-peer)# voice-class sip asymmetric payload full
```

Related Commands

Command	Description
dial-peer voice	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer configuration mode.

voice-class sip audio forced

To allow only audio and image (for T.38 Fax) media types, and drop all other media types (such as video and application), use the **voice-class sip audio forced** command in dial-peer configuration mode. To disable, use **no** form of this command.

voice-class sip audio forced [system]
no voice-class sip audio forced

Syntax Description

system	(Optional) Uses the global configuration settings to allow only audio and image (for T.38 Fax) media types.

Command Default

Support for audio forced at dial-peer level uses the global configuration level settings.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Release	Modification
Cisco IOS 15.6(2)T	This command was introduced.
Cisco IOS XE Denali 16.3.1	This command was integrated into Cisco IOS XE Denali 16.3.1.

Usage Guidelines

Use **voice-class sip audio forced** command on the dial-peer when a particular remote end does not support receiving any video or application m-lines in SDP.

Example

```
Router> enable
 Router# configure terminal
 Router(config)# dial-peer voice 1 voip
 Router(config-dial-peer)# voice-class sip audio forced
```

voice-class sip authenticate redirecting-number

To supersede global settings and enable a dial peer on a Cisco IOS voice gateway to authenticate and pass Session Initiation Protocol (SIP) credentials based on the redirecting number of forwarded calls, use the **voice-class sip authenticate redirecting-number** command in dial peer voice configuration mode. To supersede global settings and specify that a dial peer uses only the calling number of forwarded calls, use the **no** form of this command. To return a dial peer to the default setting so that the dial peer uses the global setting, use the **default** form of this command.

voice-class sip authenticate redirecting-number [**system**]
no voice-class sip authenticate redirecting-number
default voice-class sip authenticate redirecting-number

Syntax Description

system	(Optional) Specifies that the dial peer use whatever setting is configured at the global (voice service SIP) command level (default).

Command Default

The dial peer uses the global setting. If the global setting is not specifically configured, the dial peer uses only the calling number of a forwarded call for SIP credentials even when the redirecting number is available for that call.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.4(24)T	This command was introduced.

Usage Guidelines

When an INVITE message sent out by the gateway is challenged, it must respond with the appropriate SIP credentials before the call is established. The default global behavior for the gateway is to authenticate and pass SIP credentials based on the calling number and all dial peers on a gateway default to the global setting. However, for forwarded calls, it is sometimes more appropriate to use the redirecting number and this can be specified at either the global or dial peer level (configuring behavior for a specific dial peer supersedes the global setting).

Use the **voice-class sip authenticate redirecting-number** command in dial peer voice configuration mode to supersede global settings and enable a dial peer to authenticate and pass SIP credentials based on the redirecting number when available. Use the **no** form of this command to supersede global settings and force a dial peer to authenticate and pass SIP credentials based only on the calling number of forwarded calls. Use the **default** form of this command to configure the dial peer to use the global setting.

The redirecting number is present only in the headers of forwarded calls. When the **voice-class sip authenticate redirecting-number** command is disabled or the redirecting number is not available, the dial peer passes SIP credentials that are based on the calling number of the forwarded call. This is also the behavior on dial peers that are configured to use the global setting and the global setting is disabled (default). To enable the global setting (which is used as the default setting for all dial peers on the gateway), use the **authenticate redirecting-number** command in voice service SIP configuration mode.

Examples

The following example shows how to enable dial peer 2 to authenticate and pass SIP credentials based on the redirecting number (if available) of a forwarded call when a SIP INVITE message is challenged:

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2 voip

Router(config-dial-peer)# voice-class sip authenticate redirecting-number
```

The following example shows how to force dial peer 2 to authenticate and pass only the calling number of a call even when the global setting is enabled and a redirecting number is available for a call:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer)# no voice-class sip authenticate redirecting-number
```

The following two examples show different ways of setting dial peer 2 to the default setting so that it authenticates and passes either the redirecting or calling number of a call based on the global (system) setting for the gateway:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer)# default voice-class sip authenticate redirecting-number
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer)# voice-class sip authenticate redirecting-number system
```

Related Commands

Command	Description
authenticate redirecting-number	Enables a Cisco IOS voice gateway to authenticate and pass SIP credentials based on the redirecting number when available instead of the calling number of a forwarded call.

voice-class sip bind

To bind the source address of a specific interface for a dial-peer on a Session Initiation Protocol (SIP) trunk, use the **voice-class sip bind** command in dial peer voice configuration mode. To disable bind at the dial-peer level or restore the bind to the global level, use the **no** form of this command.

voice-class sip bind { **control** | **media** | **all** } **source-interface** *interface-id* [**ipv6-address** *ipv6-address*]
no voice-class sip bind { **control** | **media** | **all** }

Syntax Description

control	Binds Session Initiation Protocol (SIP) signaling packets.
media	Binds only media packets.
all	Binds SIP signaling and media packets.
source interface *interface-id*	Specifies an interface as the source address of SIP packets.
ipv6-address *ipv6-address*	(Optional) Configures the IPv6 address of the interface.

Command Default

Bind is disabled.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
15.1(2)T	This command was introduced.
Cisco IOS XE Amsterdam 17.3.1a	Introduced support for YANG models.

Usage Guidelines

Use the **voice-class sip bind** command in dial peer voice configuration mode to bind the source address for signaling and media packets to the IP address of an interface on Cisco IOS voice gateway.

You can configure multiple IPv6 addresses for an interface and select one address using the ipv6-address keyword.

Examples

The following example shows how to configure SIP bind command:

```
Router(config)# dial-peer voice 101 voip
Router(config-dial-peer)# session protocol sipv2
Router(config-dial-peer)# voice-class sip bind control source-interface GigabitEthernet0/0
 ipv6-address 2001:0DB8:0:1::1
Router(config-dial-peer)# voice-class sip bind media source-interface GigabitEthernet0/0
Router(config-dial-peer)# voice-class sip bind all source-interface GigabitEthernet0/0
```

voice-class sip block

To configure an individual dial peer on a Cisco IOS voice gateway or Cisco Unified Border Element (Cisco UBE) to drop (not pass) specific incoming Session Initiation Protocol (SIP) provisional response messages, use the **voice-class sip block** command in dial peer voice configuration mode. To disable a configuration to drop incoming SIP provisional response messages on an individual dial peer, use the **no** form of this command.

voice-class sip block {**180** | **181** | **183**} [{**sdp** {**absent** | **present**} | **system**}]
no voice-class sip block {**180** | **181** | **183**}

Syntax Description

180	Specifies that incoming SIP 180 Ringing messages should be dropped (not passed to the other leg).
181	Specifies that incoming SIP 181 Call is Being Forwarded messages should be dropped (not passed to the other leg).
183	Specifies that incoming SIP 183 Session in Progress messages should be dropped (not passed to the other leg).
sdp	(Optional) Specifies that either the presence or absence of Session Description Protocol (SDP) information in the received response determines when the dropping of specified incoming SIP messages takes place.
absent	Configures the SDP option so that specified incoming SIP messages are dropped only if SDP is absent from the received provisional response.
present	Configures the SDP option so that specified incoming SIP messages are dropped only if SDP is present in the received provisional response.
system	Configures the dial peer to use global configuration settings for dropping incoming SIP provisional response messages.

Command Default

Defaults to the global configuration setting, which, when not specifically configured, means incoming SIP 180, 181, and 183 provisional responses are forwarded.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.4(22)YB	This command was introduced. Only SIP 180 and SIP 183 messages are supported on Cisco UBEs.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.0(1)XA	This command was modified. Support was added for SIP 181 messages on the Cisco IOS SIP gateway, SIP-SIP Cisco UBEs, and the SIP trunk of Cisco Unified Communications Manager Express (Cisco Unified CME).
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.

Release	Modification
Cisco IOS XE Release 3.1S	This command was integrated into Cisco IOS XE Release 3.1S.

Usage Guidelines

Use the **voice-class sip block** command in dial peer voice configuration mode to configure a specific dial peer on a Cisco IOS voice gateway or Cisco UBE to override global settings and drop specified SIP provisional response messages. Additionally, you can use the **sdp** keyword to further control when the specified SIP message is dropped based on either the absence or presence of SDP information.

You can also use the **system** keyword to configure a specific dial peer to use global configuration settings for dropping incoming SIP provisional response messages. To configure global settings on a Cisco IOS voice gateway or Cisco UBE, use the **block** command in voice service SIP configuration mode. To disable configurations for dropping specified incoming SIP messages on an individual dial peer, use the **no voice-class sip block** command in dial peer voice configuration mode.

Note This command is supported only on outbound dial peers--it is nonoperational if configured on inbound dial peers. You should configure this command on the outbound SIP leg that sends out the initial INVITE message. Additionally, this feature applies only to SIP-to-SIP calls and will have no effect on H.323-to-SIP calls.

Examples

The following example shows how to configure dial peer 1 to override any global configurations and drop specified incoming SIP provisional response messages regardless whether SDP is present:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip block 181
```

The following example shows how to configure dial peer 1 to override any global configurations and drop specified incoming SIP provisional response messages only if SDP is present:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip block 183 sdp present
```

The following example shows how to configure dial peer 1 to override any global configurations and drop incoming SIP provisional response messages only when SDP is not present:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip block 180 sdp absent
```

The following example shows how to configure a dial peer to use the global configuration settings for dropping incoming SIP provisional response messages:

```
Router> enable
Router# configure
 terminal
```

```
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip block 181 system
```

The following example shows how to configure a dial peer to pass all incoming SIP provisional response messages regardless of global configuration settings:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# no voice-class sip block 180
```

Related Commands

Command	Description
block	Configures global configuration for dropping specified SIP provisional response messages on a Cisco IOS voice gateway or Cisco UBE.
map resp-code	Configures global settings on a Cisco UBE for mapping specific incoming SIP provisional response messages to a different SIP response message.
voice-class sip map resp-code	Configures a specific dial peer on a Cisco UBE to map specific incoming SIP provisional response messages to a different SIP response message.

voice-class sip call-route

To enable call routing based on the Destination-Route-String, P-called-party-id and History-Info header values at the dial-peer configuration level, use the **voice-class sip call-route** command in dial peer voice configuration mode. To disable Header-based routing, use the **no** form of this command.

voice-class sip call-route {**dest-route-string** | **p-called-party-id** | **history-info** | **url**} [**system**]
no voice-class sip call-route {**dest-route-string** | **p-called-party-id** | **history-info** | **url**}

Syntax Description

dest-route-string	Enables call routing based on the Destination-Route-String.
p-called-party-id	Enables call routing based on the P-Called-Party-Id header.
history-info	Enables call routing based on the History-Info header.
url	Enables call routing based on the URL.
system	(Optional) Uses the global configuration settings to enable call routing based on the header values on this dial peer.

Command Default

Support for call routing based on the Destination-Route-String, P-Called-Party-Id, History-Info headers and URL at the dial peer level is disabled.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.4(22)YB	This command was introduced.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.1(2)T	This command was modified. The **history-info** keyword was added.
15.2(1)T	This command was modified. The **url** keyword was added.
15.3(3)M	This command was modified. The **dest-route-string** keyword was added.
Cisco IOS XE Release 3.10S	This command was integrated into Cisco IOS XE Release 3.10S.

Usage Guidelines

Use the **voice-class sip call-route** command on the inbound dial peer to enable the gateway to route calls based on the received header in a received INVITE message.

The **voice-class sip call-route** command takes precedence over the **call-route** command in voice service VoIP SIP configuration mode. However, if the **voice-class sip call-route** command is used with the **system** keyword, the gateway uses the settings configured globally by the **call-route** command.

If multiple call routes are configured, call routing enabled based on destination route string takes precedence over other header configurations. Destination route string configuration is applicable only for outbound dial-peer matching.

Examples

The following example shows how to enable call routing based on the Destination-Route-String, P-Called-Party-Id, History-Info header values and URL at the dial peer configuration level:

```
Device> enable
Device# configure terminal
Device(config)# dial-peer voice 2611 voip
Device(config-dial-peer)# voice-class sip call-route dest-route-string
Device(config-dial-peer)# voice-class sip call-route p-called-party-id
Device(config-dial-peer)# voice-class sip call-route history-info
Device(config-dial-peer)# voice-class sip call-route url
```

Related Commands

Command	Description
call-route	Enables call routing based on the Destination-Route-String, P-Called-Party-Id and History-Info header values at the global configuration level.

voice-class sip calltype-video

To configure the bearer capability setting on an H.320 dial peer so that it supports unrestricted digital media, use the **voice-class sip calltype-video** command in dial peer voice configuration mode. To return the bearer capability setting for an H.320 dial peer to the default, use the **no** form of this command.

voice-class sip calltype-video
no voice-class sip calltype-video

Syntax Description

This command has no arguments or keywords.

Command Default

Bearer capability setting for support of unrestricted digital media support is disabled.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.4(24)T	This command was introduced.

Usage Guidelines

H.320 dial peers support only voice calls by default. Use the **voice-class sip calltype-video** command to configure the bearer capability setting, which enables support of unrestricted digital media calls on an H.320 dial peer.

Examples

The following example shows how to configure the bearer capability setting on dial peer 2 so that it supports unrestricted digital media:

```
Router> enable

Router# configure
 terminal
Router(config)# dial-peer voice 2 voip

Router(config-dial-peer)# voice-class sip call-type video
```

voice-class sip content sdp version increment

To increment the SDP version for any RE-INVITE with SDP change even if the previous offer sent by CUBE was rejected, use **voice-class sip content sdp version increment** command in dial-peer configuration mode.

voice-class sip content sdp version increment {**system**}

system	Uses the system level configuration for sdp version increment

Command Default

SDP version will not be incremented for any RE-INVITE with SDP change even if the previous offer sent by CUBE was rejected.

Command Modes

dial-peer configuration mode (config-dial-peer)

Command History

Release	Modification
Cisco IOS 15.5(2)T Cisco IOS XE 3.15	This command was introduced.

Usage Guidelines

Use **voice-class sip content sdp version increment** command to increment the SDP version for any RE-INVITE with SDP change even if the previous offer sent by CUBE was rejected.

Example

```
Device> enable
Device# configure terminal
Device(config)# dial-peer voice 1 voip
Device(config-dial-peer)# voice-class sip content sdp version increment
```

voice-class sip copy-list

To configure a list of entities to be sent to the peer call leg on a dial peer, use the **voice-class sip copy-list**command in dial peer configuration mode. To disable the configuration, use the **no** form of this command.

voice-class sip copy-list {*tag* | **system**}
no voice-class sip copy-list

Syntax Description

tag	Tag number of the Session Initiation Protocol (SIP) copy list. The range is from 1 to 10000.
system	Specifies to use the global level configuration to copy the list.

Command Default

Entries configured at the global level are sent to the peer call leg.

Command Modes

Dial peer configuration (config-dial-peer)

Voice class tenant

Command History

Release	Modification
15.1(3)T	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use the **voice-class sip copy-list** command to configure Cisco Unified Border Element (UBE) to pass an unsupported parameter present in a mandatory header from one peer call leg to another. You can copy the inbound message headers into variables and pass the headers to the outbound peer call leg.

Examples

The following example shows how to configure a SIP list to be sent to the peer call leg:

```
Router(config)# dial-peer voice 66 voip
Router(config-dial-peer)# voice-class sip copy-list 4
```

Related Commands

Command	Description
voice class sip-copylist	Configures a list of entities to be sent to the peer call leg.

voice-class sip e911

To enable SIP E911 system services on a dial peer, use the **voice-class sip e911**command in VoIP dialpeer configuration mode. To disable SIP E911 services, use the **no** form of this command.

voice-class sip e911
no voice-class sip e911

Syntax Description

This command has no arguments or keywords.

Command Default

The dial peer uses the global setting.

Command Modes

VoIP dialpeer configuration mode.

Command History

Release	Modification
12.4(9)T	This command was introduced.

Usage Guidelines

The **no** form of this command sets the dial peer configuration to disable, which indicates that E911 will not be used for this peer. Because the **no** version of the command causes non default behavior, it can been seen in the **show running-config** output. See also the **voice service voip sip e911** and **debug csm neat** commands.

Examples

The following examples enable and disable E911 services on a VoIP dial peer:

```
Router(config)# dial-peer voice 2
Router(config-dial-peer)# voice-class sip e911
*Jun 06 00:47:20.611: setting peer 2 to enable
Router(config-dial-peer)# no voice-class sip e911

*Jun 06 00:49:58.931: setting peer 2 to disable
```

Related Commands

Command	Description
debug csm neat	Turns on debugging for all Call Switching Module (CSM) Voice over IP (VoIP) calls.
show running-config	Displays the running configuration.
e911	Enables E911 system services for SIP voice service VoIP.

voice-class sip-event-list

To configure lists of SIP events to be passed through. To disable this feature, use the **no** form of this command.

voice-class sip-event-list *tag*
no voice-class sip-event-list *tag*

Syntax Description

tag	Event list tag. Range is 1-10000.

Command Default

No default value.

Command Modes

Global configuration (config).

Voice class tenant configuration (config-class).

Command History

Release	Modification
Cisco IOS XE 3.11S	The command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

Usage Guidelines

Use this command for troubleshooting to determine which SIP event was configured as passed through. To set the sip event, use the **voice class sip-event-list** command.

To set the voice class sip-event-list in the dial peer, use the **voice-class sip pass-thru subscribe-notify-events <event id | all>** command.

To set the voice class sip-event-list in the voice service VoIP under sip, use the **pass-thru subscribe-notify-events <event id | all>** command.

To set the voice class sip-event-list in the voice class tenant, use the **pass-thru subscribe-notify-events <event id | all>** command.

Examples

```
Router#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)#voice class sip-event-list 12
Router(config-class)#event ?
  WORD  name of event to be added in event list
Router(config-class)#event sipevent1
Router(config-class)#event sipevent2
```

Related Commands

Command	Description

voice-class sip early-media update block

To block the UPDATE requests with SDP in an early dialog, use **voice-class sip early-media update block** command in dial-peer configuration mode. To disable, use **no** form of this command.

voice-class sip early-media update block [{**re-negotiate**}]
no voice-class sip early-media update block [{**re-negotiate**}]

Syntax Description

re-negotiate	Enables end to end renegotiation if the UPDATE request contains changes in caller ID, transcoder addition or deletion, or video escalation or de-escalation.

Command Default

CUBE allows pass-through of early dialog UPDATE requests from one user agent to the other.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
Cisco IOS 15.5(3)M, Cisco IOS-XE 3.16S	This command was introduced.

Usage Guidelines

Use **voice-class sip early-media update block** command on the dial-peer where you want to block the Early Dialog UPDATE requests.

Use **re-negotiate** keyword to enable end to end renegotiation if the UPDATE request contains changes in caller ID, transcoder addition or deletion, or video escalation or de-escalation.

Examples

The following example shows early dialog update block being configured in dial-peer configuration mode:

```
Router(config-dial-peer)# voice-class sip early-media update block
```

voice-class sip encap clear-channel

To enable RFC 4040-based clear-channel codec negotiation for Session Initiation Protocol (SIP) calls on an individual dial peer, overriding the global setting on a Cisco IOS voice gateway or Cisco Unified Border Element (Cisco UBE), use the **voice-class sip encap clear-channel** command in dial peer voice configuration mode. To disable RFC 4040-based clear-channel codec negotiation on an individual dial peer for SIP calls on a Cisco IOS voice gateway or Cisco UBE, use the **no** form of this command.

voice-class sip encap clear-channel [{**standard** | **system**}]
no voice-class sip encap clear-channel standard

Syntax Description

standard	(Optional) Specifies standard RFC 4040 encapsulation.
system	(Optional) Configures the dial peer to use global configuration settings for clear-channel codec negotiation.

Command Default

The dial peer uses the system configuration. (If the global **encap clear-channel standard** command is not enabled, then legacy encapsulation [X-CCD/8000] is used for clear-channel codec negotiation.)

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
15.0(1)XA	This command was introduced.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use the **voice-class sip encap clear-channel standard** command in dial peer voice configuration mode to override global settings for clear-channel codec negotiation on a Cisco IOS voice gateway or Cisco UBE and enable RFC 4040-based clear-channel codec negotiation [CLEARMODE/8000] for SIP calls on a specific dial peer. RFC 4040-based clear-channel codec negotiation allows dial peers on Cisco IOS voice gateways and Cisco UBEs to successfully interoperate with third-party SIP gateways that do not support legacy Cisco IOS clear-channel codec encapsulation [X-CCD/8000].

When the **voice-class sip encap clear-channel standard** command is enabled on a specific dial peer on a Cisco IOS voice gateway or Cisco UBE, SIP calls on that dial peer that use the Cisco IOS clear channel codec are translated into calls that use [CLEARMODE/8000] regardless of the global configuration so that the calls do not get rejected when they reach third-party SIP gateways.

You can also use the **voice-class sip encap clear-channel system** command to configure a specific dial peer to use global configuration settings for clear-channel codec negotiation. To enable RFC 4040 clear-channel codec negotiation for SIP calls globally on a Cisco IOS voice gateway or Cisco UBE, use the **encap clear-channel standard** command in voice service SIP configuration mode. To override global settings and disable RFC 4040-based clear-channel codec negotiation on a specific dial peer, use the **no voice-class sip encap clear-channel standard** command in dial peer voice configuration mode.

Examples

The following example shows how to configure dial peer 1 to override any global configurations and enable RFC 4040-based clear-channel codec negotiation for SIP calls:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip encap clear-channel standard
```

The following example shows how to configure dial peer 1 to use the global configuration for clear-channel codec negotiation for SIP calls:

```
Router> enable
Router# configure
 terminal
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# voice-class sip encap clear-channel system
```

Related Commands

Command	Description
encap clear-channel standard	Enables RFC 4040-based clear-channel codec negotiation for SIP calls globally on a Cisco IOS voice gateway or Cisco UBE.