



## Cisco IOS Voice Commands: V

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

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

# vad (dial peer)

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

**vad [aggressive]**

**no vad [aggressive]**

## Syntax Description

<b>aggressive</b>	Reduces noise threshold from –78 to –62 dBm. Available only when session protocol multicast is configured.
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## 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 Cisco 3600 series.
12.0(4)T	This command was implemented as a dial-peer command on Cisco MC3810 (in prior releases, the <b>vad</b> command was available only as a voice-port command).
12.2(11)T	The <b>aggressive</b> keyword was added.

## Usage Guidelines

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

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

## 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
	<b>comfort-noise</b>	Generates background noise to fill silent gaps during calls if VAD is activated.
	<b>dial-peer voice</b>	Enters dial peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
	<b>vad (voice-port)</b>	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**

<b>Syntax Description</b>	<b>on</b>	Enables the local vad settings irrespective of the external vad settings.
	<b>off</b>	Disables the local vad settings irrespective of the external vad settings.
	<b>override</b>	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)

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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
<b>dsp services dspfarm</b>	Enables the DSP-farm services.
<b>dspfarm profile</b>	Enters the DSP farm profile configuration mode, and defines a profile for the DSP farm services.
<b>show dspfarm (SPA-DSP)</b>	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	
<b>maximum</b>	Sets the maximum playout buffer delay, in milliseconds (ms). Range: 40 to 1000. Default: 1000.
<i>milliseconds</i>	Delay time, in milliseconds (ms).
<b>minimum</b>	Sets the minimum playout buffer delay, in ms. Range: 10 to 40. Default: 40.
<b>mode</b>	Configures voice-band-detection playout buffer adaptation mode.
<b>fixed</b>	Sets the jitter buffer to a constant delay.
<b>no-timestamps</b>	(Optional) Fixes the jitter buffer at a constant delay without time stamps.
<b>passthrough</b>	Sets the jitter buffer passthrough mode for clock compensation.
<b>nominal</b>	Sets the nominal playout buffer delay, in ms. Range: 10 to 1000. Default: 60.

**Defaults** 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. <ul style="list-style-type: none"> <li>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.</li> <li>The maximum time value was decreased from 1700 to 1000 ms and the default of 200 was increased to 1000 ms.</li> <li>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.</li> </ul>
	12.4(24)T6	This command was modified. The <b>no-timestamps</b> keyword was added and <b>passthrough</b> keyword usage guidelines were clarified.

**Usage Guidelines** Use this command 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 **passthrough** 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
```

**Related Commands**

Command	Description
<b>voice-service</b>	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

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

## Command Default

No real-time VBR settings are configured

## Command Modes

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

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

For ATM VC bundle members: Bundle-vc configuration

## Command History

Release	Modification
12.0	This command was introduced on 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.

## 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
<b>encapsulation aal5</b>	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**

<b>Syntax Description</b>	<i>pvc-identifier</i>	Identifier for the PVC. Range is from 0 to 32767. There is no default value.
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<b>Command Default</b>	No default behavior or values
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<b>Command Modes</b>	ATM virtual circuit configuration mode
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<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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.

<b>Usage Guidelines</b>	The <i>pvc-identifier argument</i> 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 <i>pvc-identifier</i> .
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<b>Examples</b>	The following example shows how to assign a PVC identifier:
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```
Router(config-if-atm-vc)# vcci 5278
```

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>mgcp</b>	Starts the MGCP daemon.
	<b>pvc</b>	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	<b>h261</b>	Video codec H.261
	<b>h263</b>	Video codec H.263
	<b>h263+</b>	Video codec H.263+
	<b>h264</b>	Video codec H.264

<b>Command Default</b>	No video codec is configured.
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<b>Command Modes</b>	Dial peer configuration
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Command History	Release	Modification
	12.4(11)T	This command was introduced.

<b>Usage Guidelines</b>	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.
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<b>Examples</b>	The following example shows configuration for video codec H.263+ on VoIP dial peer 30:
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```
dial-peer voice 30 voip
  video codec h263+
```

Related Commands	Command	Description
	<b>video codec (voice-class)</b>	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

<b>h261</b>	Apply this preference to video codec H.261
<b>h263</b>	Apply this preference to video codec H.263
<b>h263+</b>	Apply this preference to video codec H.263+
<b>h264</b>	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.

## 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
```

## Related Commands

Command	Description
<b>video codec (dial peer)</b>	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. To disable transcoding and transsizing, use **no** form of this command.

**video screening**

**no video screening**

## Syntax Description

This command has no arguments or keywords.

## Command Default

Video screening is disabled.

## Command Modes

Voice service SIP configuration.

## Command History

Release	Modification
15.1(4)M	The command was introduced.

## 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
```

## Related Commands

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

# 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

<b>dc-voltage</b>	DC voltage VMWI is enabled on this FXS port.
<b>fsk</b>	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
<b>stcapp</b>	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		
<b>data</b>		(Required for Cisco-trunk permanent calls. Optional for switched calls.) Selects a subchannel (CID) for data other than the default subchannel, which is 4.
<i>cid</i>		(Optional) Specifies the subchannel to be used for data. Range is from 4 to 255. The default is 4. If <b>data</b> is specified, enter a valid CID.
<b>call-control</b>		(Optional) Reserves a subchannel for call-control signaling.
<b>cisco</b>		(Optional) Cisco proprietary voice encapsulation for VoFR with data is carried on CID 4 and call-control on CID 5.
<i>cid</i>		(Optional) Specifies the subchannel to be used for call-control signaling. Valid range is from 4 to 255. The default is 5. If <b>call-control</b> is specified and a CID is not entered, the default CID is used.

<b>Command Default</b>	Disabled
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<b>Command Modes</b>	Frame relay DLCI configuration
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**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.
12.0(7)XK	The use of the <b>cisco</b> option was modified. Beginning in this release, use the <b>cisco</b> 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**

Table 244 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 244**      *Combinations of the vofr Command*

Type of Call	Command Combination to Use
Switched call (user dialed or auto-ringdown) to other routers supporting VoFR	<b>vofr [data cid] [call-control [cid]]</b> <sup>1</sup>
Cisco-trunk permanent call (private-line) to other routers supporting VoFR	<b>vofr data cid call-control cid</b>
FRF.11 trunk call (private-line) to other routers supporting VoFR	<b>vofr [data cid] [call-control cid]</b> <sup>2</sup>

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
<b>class</b>	Assigns a VC class to a PVC.
<b>frame-relay interface-dlci</b>	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
	<b>resource-pool enable</b>	Enables resource pool management.
	<b>resource-pool profile service voip</b>	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*}

<b>Syntax Description</b>	<i>name</i> Designates which voicecaps to use on this voice port.				
<b>Command Default</b>	No default values or behavior				
<b>Command Modes</b>	Voice-port configuration				
<b>Command History</b>	<table><tr><th>Release</th><th>Modification</th></tr><tr><td>12.3(4)T</td><td>This command was introduced.</td></tr></table>	Release	Modification	12.3(4)T	This command was introduced.
Release	Modification				
12.3(4)T	This command was introduced.				
<b>Usage Guidelines</b>	The character value for the <i>name</i> argument must be identical to the value entered when you created the voicecap using the <b>voicecap entry</b> command.				
<b>Examples</b>	<p>The following example configures a voicecap with the name qualityERL:</p> <pre>Router&gt; enable Router# configure terminal Router(config)# voicecap entry qualityERL v270=120 Router(config)# voice-port 3/0:D Router(config-voiceport)# voicecap configure qualityERL</pre>				
<b>Related Commands</b>	<table><tr><th>Command</th><th>Description</th></tr><tr><td><b>voicecap entry</b></td><td>Creates a voicecap on NextPort platforms.</td></tr></table>	Command	Description	<b>voicecap entry</b>	Creates a voicecap on NextPort platforms.
Command	Description				
<b>voicecap entry</b>	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

<i>name string</i>	(Optional) A word and a string of characters that uniquely identify a voicecap. <ul style="list-style-type: none"> <li>The <i>name</i> argument specifies a unique identifier for a voicecap.</li> <li>The <i>string</i> argument specifies one or more voicecap register entries, similar to a modemcap. Each entry is of the form <b>vindex=value</b>, where <i>index</i> refers to a specific V register, and <i>value</i> designates the value for that V register.</li> </ul>
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## 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 [Table 245](#).

**Table 245**      **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.
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
<b>voicecap configure</b>	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

<b>carrier</b>	Carrier code address family
<b>trunk-group</b>	Trunk group address family
<b>prefix</b>	E.164 prefix
<i>seconds</i>	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 <b>capacity update interval</b> .

## 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
<b>capacity update interval (dial peer)</b>	Changes the capacity update for prefixes associated with a dial peer.
<b>capacity update interval (trunk group)</b>	Change the capacity update for carriers or trunk groups.
<b>voice call capacity stw</b>	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

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

## Defaults

The capacity reporting function is turned off.

## Command Modes

Global configuration.

## Command History

Release	Modification
12.3(1)	This command was introduced.

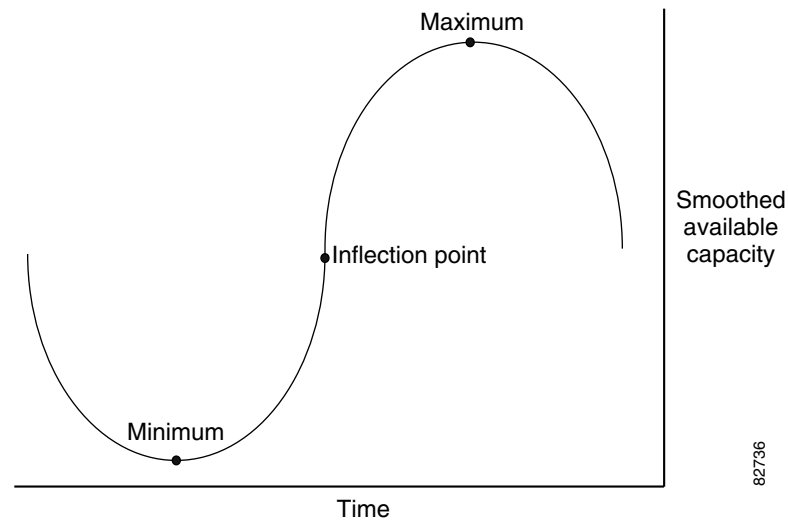
## Usage Guidelines

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 [Figure 7](#).

**Figure 7** 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
<b>voice call capacity mir</b>	Sets the values for the minimum interval between reporting (MIR) and smoothing transition time for weight (STW).
<b>voice call capacity timer interval</b>	Sets the periodic interval for reporting capacity from carrier, trunk group, or prefix databases
<b>voice call trigger hwm</b>	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 seconds**

## Syntax Description

<b>carrier</b>	Carrier code address family
<b>trunk-group</b>	Trunk group address family
<b>prefix</b>	E.164 prefix
<i>seconds</i>	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
<b>capacity update interval (dial peer)</b>	Changes the capacity update for prefixes associated with a dial peer.
<b>capacity update interval (trunk group)</b>	Change the capacity update for carriers or trunk groups.
<b>voice call capacity mir</b>	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

<b>carrier</b>	Carrier code address family
<b>trunk-group</b>	Trunk group address family
<b>prefix</b>	E.164 prefix
<i>seconds</i>	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
<b>voice call capacity mir</b>	Sets the values for the MIR and STW.
<b>voice call capacity reporting</b>	Turns on the reporting of maxima (first derivative) or inflection (second derivative) points in available capacity.
<b>voice call trigger hwm</b>	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

<b>tunnel-IEs</b>	(Optional) Information elements (IEs) are carried in the progress message.
<b>always</b>	(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 <b>tunnel-IEs</b> keyword was added.
12.3(4)XQ	The <b>always</b> keyword with the <b>tunnel-IEs</b> keyword were added.
12.3(8)T	The <b>always</b> keyword with the <b>tunnel-IEs</b> keyword were added.
12.3(9)	The <b>always</b> keyword with the <b>tunnel-IEs</b> 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
	<b>disc_pi_off</b>	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	<b>carrier</b>	Carrier code address family
	<b>trunk-group</b>	Trunk group address family
	<b>prefix</b>	E.164 prefix
	<i>value</i>	Value from 10 to 50 data points. Default is 30 data points.

Command Default	30 data points
-----------------	----------------

Command Modes	Global configuration
---------------	----------------------

Command History	<b>Release</b>	<b>Modification</b>
	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	<b>Command</b>	<b>Description</b>
	<b>voice call csr recording interval</b>	Sets the recording interval for CSR.
	<b>voice call csr reporting interval</b>	Sets the reporting interval for CSR.

# voice call csr recording interval

To set the recording interval for call success rates (CSR), use the **voice call csr recording interval** command in global configuration mode. To disable the 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

<b>carrier</b>	Carrier code address family.
<b>trunk-group</b>	Trunk group address family.
<b>prefix</b>	E.164 prefix.
<i>minutes</i>	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
<b>voice call csr data-points</b>	Sets the number of call success rate (CSR) data points.
<b>voice call csr reporting interval</b>	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

<b>carrier</b>	Carrier code address family.
<b>trunk-group</b>	Trunk group address family.
<b>prefix</b>	E.164 prefix.
<i>seconds</i>	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
<b>voice call csr data-points</b>	Sets the number of CSR data points.
<b>voice call csr recording interval</b>	Sets the recording interval for CSR.

# voice call debug

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

**voice call debug full-guid | short-header**

**no voice call debug full-guid | short-header**

Syntax Description	full-guid	Displays the GUID in a 16-byte header.
	<b>Note</b>	When the <b>no</b> version of this command is input with the <b>full-guid</b> 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 argument was removed and the short-header argument was added.

**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_fr
om_cdb:
00:05:12: //1/0E2C8A90-BC00-11D5-8002-DACCFDCEF87D/VTSP:(0:D):0:0:4385/set_playout_cdb:
00:05:12: //1/0E2C8A90-BC00-11D5-8002-DACCFDCEF87D/VTSP:(0:D):0:0:4385/vtsp_dsp_echo_cance
ller_control:
```

**Note**

The “//1/” output indicates that CallEntryID for the CCAPI module is not available.

[Table 246](#) describes significant fields shown in the display.

**Table 246** *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
<b>debug rtsp api</b>	Displays debug output for the RTSP client API.
<b>debug rtsp client session</b>	Displays debug output for the RTSP client data.
<b>debug rtsp error</b>	Displays error message for RTSP data.
<b>debug rtsp pmh</b>	Displays debug messages for the PMH.
<b>debug rtsp socket</b>	Displays debug output for the RTSP client socket data.
<b>debug voip ccapi error</b>	Traces error logs in the CCAPI.
<b>debug voip ccapi inout</b>	Traces the execution path through the CCAPI.
<b>debug voip ivr all</b>	Displays all IVR messages.
<b>debug voip ivr applib</b>	Displays IVR API libraries being processed.
<b>debug voip ivr callsetup</b>	Displays IVR call setup being processed.
<b>debug voip ivr digitcollect</b>	Displays IVR digits collected during the call.
<b>debug voip ivr dynamic</b>	Displays IVR dynamic prompt play debug.
<b>debug voip ivr error</b>	Displays IVR errors.
<b>debug voip ivr script</b>	Displays IVR script debug.
<b>debug voip ivr settlement</b>	Displays IVR settlement activities.
<b>debug voip ivr states</b>	Displays IVR states.
<b>debug voip ivr telcommands</b>	Displays the TCL commands used in the script.
<b>debug voip rawmsg</b>	Displays the raw VoIP message.
<b>debug vtsp all</b>	Enables <b>debug vtsp session</b> , <b>debug vtsp error</b> , and <b>debug vtsp dsp</b> .
<b>debug vtsp dsp</b>	Displays messages from the DSP.
<b>debug vtsp error</b>	Displays processing errors in the VTSP.
<b>debug vtsp event</b>	Displays the state of the gateway and the call events.
<b>debug vtsp port</b>	Limits VTSP debug output to a specific voice port.
<b>debug vtsp rtp</b>	Displays the voice telephony RTP packet debugging.
<b>debug vtsp send-nse</b>	Triggers the VTSP software module to send a triple redundant NSE.
<b>debug vtsp session</b>	Traces how the router interacts with the DSP.
<b>debug vtsp stats</b>	Debugs periodic statistical information sent and received from the DSP
<b>debug vtsp vofr subframe</b>	Displays the first 10 bytes of selected VoFR subframes for the interface.
<b>debug vtsp tone</b>	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
<b>disc_pi_off</b>	Enables an H.323 gateway to disconnect a call when it receives a disconnect message with a PI.
<b>voice call convert-discp- to-prog</b>	Converts a disconnect message with a PI to a progress message.

# 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
<b>progress_ind</b>	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

<i>percent</i>	The percentage deviation for trapping calls. The range of acceptable values is 1 to 100. The default is 49.
<b>vad</b>	(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 <b>no</b> 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

<b>carrier</b>	Carrier code address family
<b>trunk-group</b>	Trunk group address family
<b>prefix</b>	E.164 prefix
<i>percent</i>	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
<b>voice call capacity mir</b>	Sets the values for the minimum interval between reporting (MIR) and smoothing transition time for weight (STW).
<b>voice call capacity reporting</b>	Turns on the reporting of maxima (first derivative) or inflection (second derivative) points in available capacity.
<b>voice call capacity timer interval</b>	Sets the periodic interval for reporting capacity from carrier, trunk group, or prefix databases



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

<b>carrier</b>	Carrier code address family
<b>trunk-group</b>	Trunk group address family
<b>prefix</b>	E.164 prefix
<i>percent</i>	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
<b>voice call capacity mir</b>	Sets the values for the minimum interval between reporting (MIR) and smoothing transition time for weight (STW).
<b>voice call capacity reporting</b>	Turns on the reporting of maxima (first derivative) or inflection (second derivative) points in available capacity.
<b>voice call capacity timer interval</b>	Sets the periodic interval for reporting capacity from carrier, trunk group, or prefix databases.

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

<b>Syntax Description</b>	<b>carrier</b>	Carrier code address family
	<b>trunk-group</b>	Trunk group address family
	<b>prefix</b>	E.164 prefix
	<i>percent</i>	If <b>percent-change</b> 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 <b>lwm</b> 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 <b>hwm</b> 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.

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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**

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

<i>slot</i>	<p>Slot number for the card to be configured. The following platform-specific numbering schemes apply:</p> <ul style="list-style-type: none"> <li>• Cisco 2600 series and Cisco 2600XM: <ul style="list-style-type: none"> <li>– 0 is the Advanced Integration Module (AIM) slot in the router chassis.</li> <li>– 1 is the network module slot in the router chassis.</li> </ul> </li> <li>• Cisco 3600 series: <ul style="list-style-type: none"> <li>– A value from 1 to 6 identifies a network module slot in the router chassis.</li> </ul> </li> <li>• Cisco 3660: <ul style="list-style-type: none"> <li>– 7 is AIM slot 0 in the router chassis.</li> <li>– 8 is AIM slot 1.</li> </ul> </li> </ul>
-------------	---

## 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 <i>slot</i> 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
<b>codec complexity</b>	Matches the DSP complexity packaging to the codecs to be supported.
<b>dspfarm (voice-card)</b>	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.

## 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
<b>voice class codec</b>	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*

<b>Syntax Description</b>	<i>tag</i>	A number used to identify voice class AAA. The range is from 1 to 10000. There is no default value.
---------------------------	------------	---

<b>Command Default</b>	No default behaviors or values
------------------------	--------------------------------

<b>Command Modes</b>	Global configuration
----------------------	----------------------

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.2(11)T	This command was introduced on the Cisco 3660, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.

<b>Usage Guidelines</b>	The <b>voice class aaa</b> 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.

<b>Examples</b>	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
	<b>authentication method</b>	Specifies an authentication method for calls coming into the defined dial peer.
	<b>authorization method</b>	Specifies an authorization method for calls coming into the defined dial peer.
	<b>method</b>	Specifies an accounting method for calls coming into the defined dial peer.
	<b>accounting suppress</b>	Disables accounting that is automatically generated by the service provider module for a specific dial peer.
	<b>voice-class aaa</b>	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*

<b>Syntax Description</b>	<i>tag</i> Unique identification number assigned to one voice class. Range is 1 to 10000.
---------------------------	---

<b>Command Default</b>	No voice class is configured for busyout functions.
------------------------	---

<b>Command Modes</b>	Global configuration
----------------------	----------------------

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.1(3)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.

<b>Usage Guidelines</b>	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**.

<b>Examples</b>	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
	<b>busyout monitor ethernet</b>	Configures a voice port to monitor a local Ethernet interface for events that would trigger a voice-port busyout.
	<b>busyout monitor probe</b>	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.
	<b>busyout monitor serial</b>	Configures a voice port to monitor a serial interface for events that would trigger a voice-port busyout.
	<b>show voice busyout</b>	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	<b>inbound</b>	Inbound voice class called number.
	<b>outbound</b>	Outbound voice class called number.
	<b>pool</b>	Voice class called number pool.
	<i>tag</i>	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	<b>Release</b>	<b>Modification</b>
	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 <b>index</b> (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
	<b>show voice class called-number</b>	Displays a specific voice class called number.
	<b>voice-class called-number (dial peer)</b>	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

<i>number</i>	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
<b>voice class codec</b>	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

<i>tag</i>	Unique number that you assign to the voice class. Range is from 1 to 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.

## 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.

## Examples

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

```
voice class codec 10
```

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

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

```
voice class codec 99
 codec preference 1 g711alaw
 codec preference 2 g711ulaw bytes 80
 codec preference 3 g723ar53
```



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

**Related Commands**

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

<i>cptone-name</i>	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
<b>dualtone</b>	Defines the tone and cadence for a custom call-progress tone.
<b>supervisory custom-cptone</b>	Associates a class of custom call-progress tones with a voice port.
<b>voice class dualtone-detect-params</b>	Modifies the boundaries and limits for call-progress tones.

# voice class dualtone

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

**voice class dualtone** *tag*

**no voice class dualtone** *tag*

<b>Syntax Description</b>	<i>tag</i> Unique identification number assigned to one voice class. Range is from 1 to 10000.
---------------------------	--

<b>Command Default</b>	No voice class is configured for tone detection parameters.
------------------------	---

<b>Command Modes</b>	Global configuration
----------------------	----------------------

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.1(3)T	This command was introduced on the Cisco 2600 series, Cisco 3600, and the Cisco MC3810.

<b>Usage Guidelines</b>	Use this command first to create the voice class. Then use the <b>supervisory disconnect dualtone voice-class</b> 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).

<b>Examples</b>	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
```

## voice class dualtone

```

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
<b>supervisory disconnect dualtone voice-class</b>	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*

<b>Syntax Description</b>	<i>tag</i>	Unique tag identification number assigned to a voice class. Range is from 1 to 10000.
---------------------------	------------	---

<b>Command Default</b>	No voice class is configured for defining answer-supervision tolerance limits.
------------------------	--

<b>Command Modes</b>	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.

<b>Usage Guidelines</b>	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 <b>voice class custom-cptone</b> command. Use the <b>supervisory dualtone-detect-params</b> command to apply these tolerance parameters to a voice port.
-------------------------	--

<b>Examples</b>	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
	<b>supervisory</b>	Assigns the boundary and detection tolerance parameters defined by the
	<b>dualtone-detect-params</b>	<b>voice class dualtone-detect-params</b> command to a voice port.
	<b>voice class custom-cptone</b>	Creates a voice class for defining custom call-progress tones.

# voice class h323

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

**voice class h323** *tag*

**no voice class h323**

Syntax Description	<i>tag</i>	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, Cisco AS5300, Cisco uBR910, and Cisco uBR924.

Usage Guidelines	The <b>voice class h323</b> command in global configuration mode does not include a hyphen. The <b>voice-class h323</b> 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 <b>voice-class h323</b> 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
	<b>voice-class h323</b>	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	<i>number</i>	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.

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
	<b>voice class codec</b>	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*

<b>Syntax Description</b>	<i>tag</i> Unique number that you assign to the voice class. Range is from 1 to 10000.
---------------------------	--

<b>Command Default</b>	No voice class is configured.
------------------------	-------------------------------

<b>Command Modes</b>	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.

<b>Usage Guidelines</b>	The <b>voice class permanent</b> command can be used for Voice over Frame Relay (VoFR), Voice over ATM (VoATM), and Voice over IP (VoIP) trunks.
	The <b>voice class permanent</b> command in global configuration mode is entered without a hyphen. The <b>voice-class permanent</b> command in dial-peer and voice-port configuration modes is entered with a hyphen.

<b>Examples</b>	The following example shows how to create a permanent voice class starting from global configuration mode:
	<pre>voice class permanent 10   signal keepalive 3 exit</pre>

**Related Commands**

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

<b>Syntax Description</b>	<i>tag</i>	Unique tag to identify the resource. The range is from 1 to 5.
---------------------------	------------	--

<b>Command Default</b>	No resource groups are created.
------------------------	---------------------------------

<b>Command Modes</b>	Global configuration (config)
----------------------	-------------------------------

<b>Command History</b>	Release	Modification
	15.1(2)T	This command was introduced.

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

<b>Examples</b>	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
```

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

# 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*

<b>Syntax Description</b>	<i>tag</i>	Voice class Session Initiation Protocol (SIP) copylist tag. The range is from 1 to 10000.
---------------------------	------------	---

<b>Command Default</b>	No header is sent to the peer call leg.
------------------------	---

<b>Command Modes</b>	Global configuration (config)
----------------------	-------------------------------

Command History	Release	Modification
	15.1(3)T	This command was introduced.

<b>Usage Guidelines</b>	Use the <b>voice class sip-copylist</b> 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.
-------------------------	--

<b>Examples</b>	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
	<b>sip-header</b>	Specifies the SIP header to be sent to the peer call leg.

# 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	<i>number</i>	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)
---------------	-------------------------------

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 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
	<b>voice class codec</b>	Assigns an identification tag number for a codec voice class.

# voice-class stun-usage

To configure voice class, enter voice class configuration mode called **stun-usage** and use the **voice-class stun-usage** command in global, dial-peer, ephone, ephone template, voice register pool, or voice register pool template configuration mode. To disable the voice class, use the **no** form of this command.

**voice-class stun-usage** *tag*

**no voice-class stun-usage** *tag*

## Syntax Description

<i>tag</i>	Unique identifier in the range 1 to 10000.
------------	--

## Command Default

The voice class is not defined.

## Command Modes

Global configuration (config)  
 Dial peer configuration (config-dial-peer)  
 Ephone configuration (config-ephone)  
 Ephone template configuration (config-ephone-template)  
 Voice register pool configuration (config-register-pool)  
 Voice register pool template configuration (config-register-pool)

## Command History

Release	Cisco Product	Modification
12.4(22)T	Cisco Unified CME 7.0	This command was introduced.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. This command can be enabled in ephone summary, ephone template, voice register pool, or voice register pool template configuration mode.

## Usage Guidelines

When the voice-class stun-usage is removed, the same is removed automatically from the dial-peer, ephone, ephone template, voice register pool, or voice register pool template configurations.

## Examples

The following example shows how to set the **voice class stun-usage** tag to 10000:

```
Router(config)# voice class stun-usage 10000
Router(config-ephone)# voice class stun-usage 10000
Router(config-voice-register-pool)# voice class stun-usage 10000
```

## Related Commands

Command	Description
<b>stun usage firewall-traversal flowdata</b>	Enables firewall traversal using STUN.
<b>stun flowdata agent-id</b>	Configures the agent ID.

# 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*

<b>Syntax Description</b>	<i>tag</i>	Label that uniquely identifies the voice class. Can be up to 32 alphanumeric characters.
---------------------------	------------	--

<b>Command Default</b>	No default behavior or values
------------------------	-------------------------------

<b>Command Modes</b>	Global configuration
----------------------	----------------------

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.3(4)XD	This command was introduced.
	12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.

<b>Usage Guidelines</b>	Use the <b>voice class tone-signal</b> 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 <b>inject guard-tone</b> , <b>inject pause</b> , and <b>inject tone</b> 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.

<b>Examples</b>	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
	<b>inject guard-tone</b>	Plays out a guard tone with the voice packet.
	<b>inject pause</b>	Specifies a pause between injected tones.
	<b>inject tone</b>	Specifies a wakeup or frequency selection tone to be played out before the voice packet.
	<b>timing delay-voice tdm</b>	Specifies the delay before a voice packet is played out.
	<b>voice-class tone-signal</b>	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	<i>tag</i>	Label that uniquely identifies the voice class. Can be up to 32 alphanumeric characters.
	<b>sip</b>	Voice class for SIP URIs.
	<b>tel</b>	Voice class for TEL URIs.

Command Default	No default behavior or values
-----------------	-------------------------------

Command Modes	Global configuration
---------------	----------------------

Command History	<b>Release</b>	<b>Modification</b>
	12.3(4)T	This command was introduced.

Usage Guidelines	<ul style="list-style-type: none"><li>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.</li><li>To reference this voice class for incoming calls, use the <b>incoming uri</b> command in the inbound dial peer. To reference this voice class for outgoing calls, use the <b>destination uri</b> command in the outbound dial peer.</li><li>Using the <b>no voice class uri</b> command removes the voice class from any dial peer where it is configured with the <b>destination uri</b> or <b>incoming uri</b> commands.</li></ul>
------------------	--

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

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

Command	Description
<b>show dialplan uri</b>	Displays which outbound dial peer is matched for a specific destination URI.
<b>voice class uri</b>	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

<i>tag</i>	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
<b>voice class aaa</b>	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

<b>inbound</b>	Assigns an inbound voice class called number to the dial peer.
<b>outbound</b>	Assigns an outbound voice class called number to the dial peer.
<i>tag</i>	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
<b>voice class called number</b>	Defines a voice class called number or range of numbers for H.320 calls.
<b>voice-class called-number-pool</b>	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**

<b>Syntax Description</b>	<i>tag</i> Digits that identify a specific voice class called number pool.
---------------------------	--

<b>Command Default</b>	No voice class called number pool is assigned to the voice port.
------------------------	--

<b>Command Modes</b>	Voice class configuration
----------------------	---------------------------

<b>Command History</b>	Release	Modification
	12.4(11)T	This command was introduced.

<b>Usage Guidelines</b>	Use this command to assign a voice class called number pool to a voice port for a dynamic H.320 secondary call dial plan.
-------------------------	---

<b>Examples</b>	The following example shows configuration for voice class called number pool 100 on voice port 1/0/0: <pre>voice-port 1/0/0 voice-class called-number-pool 100</pre>
-----------------	---

<b>Related Commands</b>	Command	Description
	<b>voice class called number</b>	Defines a voice class called number or range of numbers for H.320 calls.
	<b>voice-class called-number (dial peer)</b>	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**

<b>Syntax Description</b>	<b>tag</b>	Unique number assigned to the voice class. The range is from 1 to 10000. <ul style="list-style-type: none"> <li>This tag number maps to the tag number created using the <b>voice class codec</b> command available in global configuration mode.</li> </ul>
	<b>offer-all</b>	(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)

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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 <b>offer-all</b> keyword was added.

**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**

<b>Command</b>	<b>Description</b>
<b>show dial-peer voice</b>	Displays the configuration for all dial peers configured on the router.
<b>test voice port detector</b>	Defines the order of preference in which network dial peers select codecs.
<b>voice class codec</b>	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*

<b>Syntax Description</b>	<i>tag</i> Unique number to identify the voice class. Range is from 1 to 10000.
---------------------------	---

<b>Command Default</b>	The dial peer does not use an H.323 voice class.
------------------------	--

<b>Command Modes</b>	Dial peer configuration
----------------------	-------------------------

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.1(2)T	This command was introduced.

<b>Usage Guidelines</b>	The voice class that you assign to the dial peer must be configured using the <b>voice class h323</b> 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 <b>voice-class h323</b> command in dial peer configuration mode includes a hyphen and in global configuration mode does not include a hyphen.

<b>Examples</b>	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 <b>voice-class h323</b> command to apply voice class 4 to dial peer 36.
	<pre> Router(config)# <b>voice class h323 4</b> Router(config-class)# <b>no telephony-service ccm-compatible</b> Router(config-class)# <b>exit</b> Router(config)# <b>dial-peer voice 36 voip</b> Router(config-dial-peer)# <b>destination-pattern 555....</b> Router(config-dial-peer)# <b>session target ipv4:10.5.6.7</b> Router(config-dial-peer)# <b>voice-class h323 4</b> </pre>

**Related Commands**

Command	Description
<b>show dial-peer voice</b>	Displays the configuration for all dial peers configured on the router.
<b>voice class h323</b>	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

<i>tag</i>	Unique number assigned to the voice class. The <i>tag</i> number maps to the tag number created using the <b>voice class permanent</b> 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**

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

<i>tag</i>	Unique number assigned to the voice class. The <i>tag</i> number maps to the tag number created using the <b>voice class permanent</b> 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
	<b>signal keepalive</b>	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
	<b>signal pattern</b>	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
	<b>signal timing idle suppress-voice</b>	Configures the signal timing parameter for the idle state of a call.
	<b>signal timing oos</b>	Configures the signal timing parameter for the OOS state of a call.
	<b>signal-type</b>	Sets the signaling type for a network dial peer.
	<b>voice class permanent</b>	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]**

<b>Syntax Description</b>	<b>system</b> (Optional) Configures ANAT globally.				
<b>Command Default</b>	ANAT is enabled on SIP trunks.				
<b>Command Modes</b>	SIP configuration (conf-serv-sip) Dial peer configuration (config-dial-peer)				
<b>Command History</b>	<table> <tr> <th>Release</th><th>Modification</th></tr> <tr> <td>12.4(22)T</td><td>This command was introduced.</td></tr> </table>	Release	Modification	12.4(22)T	This command was introduced.
Release	Modification				
12.4(22)T	This command was introduced.				

<b>Usage Guidelines</b>	<p>Both the Cisco IOS SIP gateway and Cisco Unified Border Element are required to support the Session Description Protocol (SDP) ANAT semantics. The <b>bind</b> 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.</p> <p>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.</p> <p>The <b>system</b> keyword configures ANAT on all network dial peers, including the local dial peer. Using the <b>voice-class sip anat</b> command without the <b>system</b> keyword enables ANAT only for the local dial peer.</p>
-------------------------	--

<b>Examples</b>	<p>The following example globally enables ANAT on a SIP trunk:</p> <pre>Router(config-serv-sip)# voice-class sip anat system</pre> <p>The following example enables ANAT on a specified dial peer:</p> <pre>Router(config-dial-peer)# voice-class sip anat</pre>
-----------------	--



**Related Commands**

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

# voice-class sip associate registered-number

To associate the preloaded route and outbound proxy details to the registered number in the dial 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**

<b>Syntax Description</b>	<i>number</i>	Registered number. The number must be between 4 and 32.
	<b>system</b>	(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)

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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
```

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>associate registered-number</b>	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

<b>dtmf</b>	Provides asymmetric support only for dual-tone multi-frequency (DTMF) payloads.
<b>dynamic-codecs</b>	Provides asymmetric support only for dynamic codec payloads.
<b>full</b>	Provides asymmetric support both for DTMF and dynamic codec payloads.
<b>system</b>	(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
<b>dial-peer voice</b>	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer configuration mode.

# 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**

<b>Syntax Description</b>	<b>system</b> (Optional) Specifies that the dial peer use whatever setting is configured at the global (voice service SIP) command level (default).	
<b>Command Default</b>	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.	
<b>Command Modes</b>	Dial peer voice configuration (config-dial-peer)	
<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.4(24)T	This command was introduced.
<b>Usage Guidelines</b>	<p>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).</p> <p>Use the <b>voice-class sip authenticate redirecting-number</b> 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 <b>no</b> 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 <b>default</b> form of this command to configure the dial peer to use the global setting.</p>	

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
<b>authenticate redirecting-number</b>	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** } **source-interface** *interface-id* [**ipv6-address** *ipv6-address*]

**no voice-class sip bind** { **control** | **media** }

## Syntax Description

<b>control</b>	Binds Session Initiation Protocol (SIP) signaling packets.
<b>media</b>	Binds only media packets.
<b>source-interface</b> <i>interface-id</i>	Specifies an interface as the source address of SIP packets.
<b>ipv6-address</b> <i>ipv6-address</i>	(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.

## 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
```

# 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		
<b>180</b>		Specifies that incoming SIP 180 Ringing messages should be dropped (not passed to the other leg).
<b>181</b>		Specifies that incoming SIP 181 Call is Being Forwarded messages should be dropped (not passed to the other leg).
<b>183</b>		Specifies that incoming SIP 183 Session in Progress messages should be dropped (not passed to the other leg).
<b>sdp</b>		(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.
<b>absent</b>		Configures the SDP option so that specified incoming SIP messages are dropped only if SDP is absent from the received provisional response.
<b>present</b>		Configures the SDP option so that specified incoming SIP messages are dropped only if SDP is present in the received provisional response.
<b>system</b>		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.

**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
<b>block</b>	Configures global configuration for dropping specified SIP provisional response messages on a Cisco IOS voice gateway or Cisco UBE.
<b>map resp-code</b>	Configures global settings on a Cisco UBE for mapping specific incoming SIP provisional response messages to a different SIP response message.
<b>voice-class sip map resp-code</b>	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 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 {p-called-party-id | history-info} [system]**

**no voice-class sip call-route {p-called-party-id | history-info}**

## Syntax Description

<b>p-called-party-id</b>	Enables call routing based on the p-called-party-id header.
<b>history-info</b>	Enables call routing based on the history-info header.
<b>system</b>	(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 p-called-party-id and history-info headers 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 <b>history-info</b> keyword was introduced.

## 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.

## Examples

The following example shows how to enable call routing based on the p-called-party-id and history-info header values, at the dial-peer configuration level:

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

**Related Commands**

Command	Description
<b>call-route</b>	Enables call routing based on the 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 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**

<b>Syntax Description</b>	<i>tag</i>	Tag number of the Session Initiation Protocol (SIP) copy list. The range is from 1 to 10000.
	<b>system</b>	Specifies to use the global level configuration to copy the list.

<b>Command Default</b>	Entries configured at the global level are sent to the peer call leg.
------------------------	---

<b>Command Modes</b>	Dial peer configuration (config-dial-peer)
----------------------	--

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	15.1(3)T	This command was introduced.

<b>Usage Guidelines</b>	Use the <b>voice-class sip copy-list</b> 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.
-------------------------	--

<b>Examples</b>	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
```

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>voice class sip-copylist</b>	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.

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 be 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
```

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

# 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	<b>standard</b>	(Optional) Specifies standard RFC 4040 encapsulation.
	<b>system</b>	(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 <b>encap clear-channel standard</b> 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
<b>encap clear-channel standard</b>	Enables RFC 4040-based clear-channel codec negotiation for SIP calls globally on a Cisco IOS voice gateway or Cisco UBE.