Cisco IOS Voice Commands: V

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

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

vad (dial peer)

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

vad [**aggressive**]

no vad [**aggressive**]

Syntax Description

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

Command Default

VAD is enabled

Aggressive VAD is enabled in multicast dial peers

Command Modes

Dial peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series.
12.0(4)T	This command was implemented as a dial-peer command on Cisco MC3810 (in prior releases, the **vad** command was available only as a voice-port command).
12.2(11)T	The **aggressive** keyword was added.

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

vbd-playout-delay maximum

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

vbd-playout-delay maximum *time*

no vbd-playout-delay maximum

Syntax Description

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

Command Default

200 milliseconds

Command Modes

Voice-service configuration

Command History

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

Examples

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

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

Related Commands

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

vbd-playout-delay minimum

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

vbd-playout-delay minimum *time*

no vbd-playout-delay minimum

Syntax Description

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

Command Default

4 milliseconds

Command Modes

Voice-service configuration

Command History

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

Examples

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

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

Related Commands

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

vbd-playout-delay mode

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

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

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

Syntax Description

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

Command Default

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

Command Modes

Voice-service configuration

Command History

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

Usage Guidelines

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

Examples

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

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

Related Commands

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

vbd-playout-delay nominal

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

vbd-playout-delay nominal *time*

no vbd-playout-delay nominal

Syntax Description

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

Command Default

100 milliseconds

Command Modes

Voice-service configuration

Command History

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

Examples

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

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

Related Commands

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

vbr-rt

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

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

no vbr-rt

Syntax Description

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

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

vcci

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

vcci *pvc-identifier*

no vcci

Syntax Description

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

Command Default

No default behavior or values

Command Modes

ATM virtual circuit configuration mode

Command History

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

Usage Guidelines

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

Examples

The following example shows how to assign a PVC identifier:

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

Related Commands

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

video codec (dial peer)

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

video codec {**h261** | **h263** | **h263+** | **h264**}

no video codec

Syntax Description

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

Command Default

No video codec is configured.

Command Modes

Dial peer configuration

Command History

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

Usage Guidelines

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

Examples

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

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

Related Commands

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

video codec (voice class)

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

video codec {**h261** | **h263** | **h263+** | **h264**}

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

Syntax Description

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

Command Default

No video codec is configured.

Command Modes

Voice class configuration

Command History

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

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
video codec (dial peer)	Specifies a video codec for a VoIP dial peer.

vofr

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

Switched Calls

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

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

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

vofr [**cisco**]

no vofr [**cisco**]

Cisco-Trunk Permanent Calls

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

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

FRF.11 Trunk Calls

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

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

Syntax Description

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

Command Default

Disabled

Command Modes

Frame relay DLCI configuration

Command History

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

Usage Guidelines

Table 239 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 239 Combinations of the vofr Command

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

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

Examples

The following example, beginning in global configuration mode, shows how to enable VoFR on serial interface 1/1, DLCI 100. The example configures CID 4 for data; no call-control CID is defined.

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

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

```
vofr call-control
```

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

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

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

```
vofr call-control 15
```

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

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

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

```
vofr data 10
```

Related Commands

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

voice

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

voice

no voice

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Service profile configuration mode

Command History

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

Examples

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

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

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

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

Related Commands

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

voice call capacity mir

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

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

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

Syntax Description

carrier	Carrier code address family
trunk-group	Trunk group address family
prefix	E.164 prefix
seconds	Minimum interval, in seconds, with a range of 1 to 3600 seconds and a default of 10. This value cannot be set higher than the time configured for the **capacity update interval**.

Command Default

10 seconds.

Command Modes

Global configuration.

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

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

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

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

Examples

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

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

Related Commands

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

voice call capacity reporting

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

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

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

Syntax Description

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

Command Default

The capacity reporting function is turned off.

Command Modes

Global configuration.

Command History

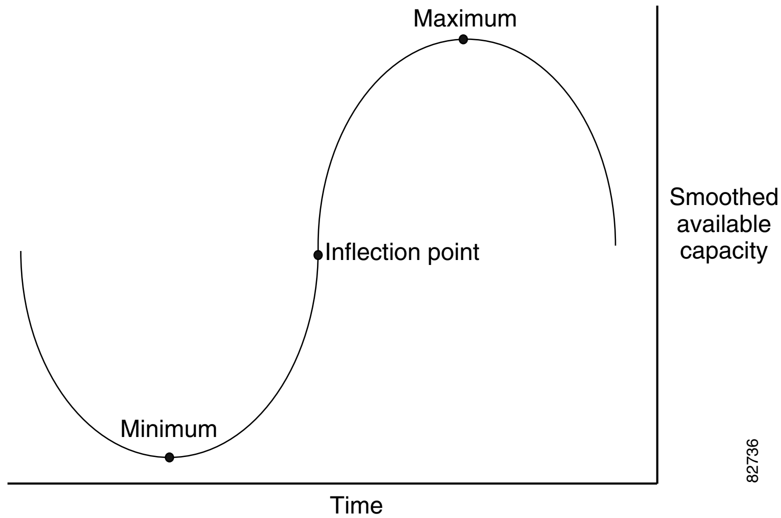
Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

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

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

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



Examples

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

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

Related Commands

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

voice call capacity 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

carrier	Carrier code address family
trunk-group	Trunk group address family
prefix	E.164 prefix
seconds	Transitions time can be from 0 to 60 seconds with a default of 10.

Command Default

10 seconds.

Command Modes

Global configuration.

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

voice call capacity timer interval

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

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

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

Syntax Description

carrier	Carrier code address family
trunk-group	Trunk group address family
prefix	E.164 prefix
seconds	Value from 10 to 3600 seconds.

Command Default

25 seconds

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

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

Examples

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

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

Related Commands

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

voice call convert-discpi-to-prog

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

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

no voice call convert-discpi-to-prog

Syntax Description

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

Command Default

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

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

Examples

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

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

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

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

Related Commands

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

voice call csr data-points

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

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

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

Syntax Description

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

Command Default

30 data points

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Examples

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

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

Related Commands

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

voice call csr recording interval

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

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

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

Syntax Description

carrier	Carrier code address family.
trunk-group	Trunk group address family.
prefix	E.164 prefix.
minutes	Value from 10 to 1000 minutes with a default of 60.

Command Default

60 minutes

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Examples

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

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

Related Commands

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

voice call csr reporting interval

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

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

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

Syntax Description

carrier	Carrier code address family.
trunk-group	Trunk group address family.
prefix	E.164 prefix.
seconds	Value from 10 to 10000 seconds with a default of 25.

Command Default

25 seconds

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Examples

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

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

Related Commands

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

voice call debug

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

voice call debug full-guid | short-header

no voice call debug full-guid | short-header

Syntax Description

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

Command Default

The short 6-byte header displays.

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	The new debug header was added to the following platforms: Cisco 2600 series, Cisco 3620, Cisco 3640, Cisco 3660 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, Cisco AS5850, and Cisco MC3810.
12.2(15)T	The header-only 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 240 describes significant fields shown in the display.

Table 240 voice call debug full-guid Field Descriptions

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

The following is sample output when the **short-header** keyword is specified:

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

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

Related Commands

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

voice call disc-pi-off

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

voice call disc-pi-off

no voice call disc-pi-off

Syntax Description

This command has no keywords or arguments.

Command Default

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

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

Examples

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

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

Related Commands

Command	Description
disc_pi_off	Enables an H.323 gateway to disconnect a call when it receives a disconnect message with a PI.
voice call convert-discpi-to-prog	Converts a disconnect message with a PI to a progress message.

voice call send-alert

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

voice call send-alert

no voice call send-alert

Syntax Description

This command has no arguments or keywords.

Command Default

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

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

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

Examples

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

```
voice call send-alert
```

Related Commands

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

voice call trigger hwm

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

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

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

Syntax Description

carrier	Carrier code address family
trunk-group	Trunk group address family
prefix	E.164 prefix
percent	Value can be 50 to 100 percent with a default of 80. If set to 100, this trigger will be turned off.

Command Default

80 percent

Command Modes

Global configuration.

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

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

Examples

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

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

Related Commands

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

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

voice call trigger lwm

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

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

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

Syntax Description

carrier	Carrier code address family
trunk-group	Trunk group address family
prefix	E.164 prefix
percent	Value can be 0 to 30 percent with a default of 10. If set to 0, this trigger will be turned off.

Command Default

10 percent

Command Modes

Global configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

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

Examples

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

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

Related Commands

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

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

voice call trigger percent-change

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

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

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

Syntax Description

carrier	Carrier code address family
trunk-group	Trunk group address family
prefix	E.164 prefix
percent	If **percent-change** is selected, value can be 0 to 100 percent with a default of 30. If set to 0, this trigger will be turned off. If **lwm** is selected, value can be 0 to 30 percent with a default of 10. If set to 0, this trigger will be turned off. If **hwm** is select, value can be 50 to 100 percent with a default of 80. If set to 100, this trigger will be turned off.

Command Default

30 percent

Command Modes

Global configuration.

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

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

Examples

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

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

Related Commands

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

voicecap configure

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

voicecap configure {*name*}

no voicecap configure {*name*}

Syntax Description

name	Designates which voicecaps to use on this voice port.

Command Default

No default values or behavior

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

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

Examples

The following example configures a voicecap with the name qualityERL:

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

Related Commands

Command	Description
voicecap entry	Creates a voicecap on NextPort platforms.

voicecap entry

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

voicecap entry [*name string*]

no voicecap entry [*name string*]

Syntax Description

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

Command Default

No voice caps can be applied to configure firmware.

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

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

Starting with Cisco IOS Release 12.4(4)XC, this command can be used to configure GSMAMR-NB codecs on Cisco AS5350XM and Cisco AS5400XM platforms. The register values for GSMAMR-NB are shown in Table 241.

Table 241 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
voicecap configure	Applies a voicecap to the specified voice ports.

voice-card

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

voice-card *slot*

Syntax Description

slot	Slot number for the card to be configured. The following platform-specific numbering schemes apply: • Cisco 2600 series and Cisco 2600XM: – 0 is the Advanced Integration Module (AIM) slot in the router chassis. – 1 is the network module slot in the router chassis. • Cisco 3600 series: – A value from 1 to 6 identifies a network module slot in the router chassis. • Cisco 3660: – 7 is AIM slot 0 in the router chassis. – 8 is AIM slot 1.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

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

- On Cisco 2600 series, Cisco 2600XM, Cisco 3660, Cisco 3725, and Cisco 3745, the *slot* argument corresponds to the physical chassis slot of the network module that has DSP resources to be configured.

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

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

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

Examples

The following example enters voice-card configuration mode to configure resources on the network module in slot 1:

```
voice-card 1
```

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

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

Related Commands

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

voice class aaa

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

voice class aaa *tag*

no voice class aaa *tag*

Syntax Description

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

Command Default

No default behaviors or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

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

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

Related Commands

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

voice-class aaa (dial peer)

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

voice-class aaa *tag*

Syntax Description

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

Command Default

No default behaviors or values

Command Modes

Dial peer configuration

Command History

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

Usage Guidelines

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

Examples

The following example shows redirecting AAA requests using Digital Number Identification Service (DNIS). You define a voice class to specify the AAA methods and then use this command.

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

Related Commands

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

voice class busyout

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

voice class busyout *tag*

no voice class busyout *tag*

Syntax Description

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

Command Default

No voice class is configured for busyout functions.

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

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

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

Related Commands

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

voice-class called-number (dial peer)

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

voice-class called-number [**inbound** | **outbound**] *tag*

no voice-class called-number

Syntax Description

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

Command Default

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

Command Modes

Dial peer configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

voice class called number

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

voice class called number {**inbound** | **outbound** | **pool**} *tag*

no voice class called number

Syntax Description

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

Command Default

No voice class called number is configured.

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

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

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

Related Commands

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

voice-class called-number-pool

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

voice-class called-number-pool *tag*

no voice-class called-number-pool

Syntax Description

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

Command Default

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

Command Modes

Voice class configuration

Command History

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

Usage Guidelines

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

Examples

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

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

Related Commands

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

voice class codec

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

voice class codec *tag*

no voice class codec *tag*

Syntax Description

tag	Unique number that you assign to the voice class. Range is 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
codec preference	Specifies a list of preferred codecs to use on a dial peer.
test voice port detector	Defines the order of preference in which network dial peers select codecs.
voice-class codec (dial peer)	Assigns a previously configured codec selection preference list to a dial peer.

voice-class codec (dial peer)

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

voice-class codec *tag*

no voice-class codec *tag*

Syntax Description

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

Command Default

Dial peers have no codec voice class assigned.

Command Modes

Dial peer configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

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

voice class custom-cptone

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

voice class custom-cptone *cptone-name*

no voice class custom-cptone *cptone-name*

Syntax Description

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

Command Default

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

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

Examples

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

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

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

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

Related Commands

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

voice class dualtone

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

voice class dualtone *tag*

no voice class dualtone *tag*

Syntax Description

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

Command Default

No voice class is configured for tone detection parameters.

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

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

```
voice class dualtone 100
```

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

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

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

Related Commands

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

voice class dualtone-detect-params

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

voice class dualtone-detect-params *tag*

no voice class dualtone-detect-params *tag*

Syntax Description

tag	Unique tag identification number assigned to a voice class. Range is from 1 to 10000.

Command Default

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

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

Examples

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

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

Related Commands

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