Enhanced ITU-T G.168 Echo Cancellation

This document describes the third-party G.168 extended echo canceller (EC) used in Cisco gateways with Cisco IOS Release 12.2(13)ZH. The extended EC uses the Cisco voice digital signal processor (DSP) code base (DSPWare).

The Enhanced ITU-T G.168 Echo Cancellation feature provides an alternative to the default Cisco-proprietary G.165 EC. The new extended EC provides improved performance for trunking gateway applications and provides a configurable tail length that supports up to 64 ms of echo cancellation.

The extended EC offers the following improvements over the Cisco default EC:

- Complies with the ITU-T G.168 (2000) standard in addition to maintaining support for the old ITU-T G.165 standard.
- Increases the configurable tail length from a maximum of 32 ms to a maximum of 64 ms.

Note Tone detection and echo disabling are performed outside the EC automatically in the DSP firmware.

Cisco IOS software supports the following improvements with the extended EC:

- Configuration and reporting of extended echo path capacity
- Configuration and reporting of worst-case echo return loss (ERL)
- Test mode support for manually freezing, thawing, and clearing the EC h-register
- Reporting of statistics for location of the largest reflector
- Reporting of the internal state of the EC

This feature provides the following additional benefits:

- No changes to platform—Improves platform functionality by updating the EC module through a DSPWare upgrade and a Cisco IOS software upgrade
- Enabling and disabling of nonlinear processor—Enables and disables nonlinear processor (NLP) spectrally matched comfort noise
- Echo return loss (ERL) configuration—Can be set to three values: 0 dB, 3 dB, and 6 dB
- Expansion of Echo Canceller Capacity—EC capacity is expanded to 64 ms

Table 1 contains specific high-complexity and medium-complexity support listed by platform. For hardware support documentation, refer to links provided in the "Related Documents" section on page 61.

Feature Specifications for Enhanced ITU-T G.168 Echo Cancellation

Feature History	
Release	**Modification**
12.2(13)T	This feature was introduced.
12.2(13)ZH	This feature replaces the Cisco G.165 EC with the extended G.168 EC as default on the Cisco 1700 series and the Cisco 7750.
Supported Platforms	
For platforms supported in Cisco IOS Release 12.2(13)ZH, consult Cisco Feature Navigator.	

Determining Platform Support Through Cisco Feature Navigator

Cisco IOS software is packaged in feature sets that are supported on specific platforms. To get updated information regarding platform support for this feature, access Cisco Feature Navigator. Cisco Feature Navigator dynamically updates the list of supported platforms as new platform support is added for the feature.

Cisco Feature Navigator is a web-based tool that enables you to determine which Cisco IOS software images support a specific set of features and which features are supported in a specific Cisco IOS image. You can search by feature or release. Under the release section, you can compare releases side by side to display both the features unique to each software release and the features in common.

To access Cisco Feature Navigator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check verifies that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions found at this URL:

http://www.cisco.com/register

Cisco Feature Navigator is updated regularly when major Cisco IOS software releases and technology releases occur. For the most current information, go to the Cisco Feature Navigator home page at the following URL:

http://www.cisco.com/go/fn

Availability of Cisco IOS Software Images

Platform support for particular Cisco IOS software releases is dependent on the availability of the software images for those platforms. Software images for some platforms may be deferred, delayed, or changed without prior notice. For updated information about platform support and availability of software images for each Cisco IOS software release, refer to the online release notes or, if supported, Cisco Feature Navigator.

Table 1 Extended Echo Canceller Algorithm Coverage by Platform

Platform	Module	High Complexity		Medium Complexity		Comments
		Analog	Digital	Analog	Digital	
Cisco 1700 series		12.2.13T, 12.2(8)YN	12.2.13T, 12.2(8)YN	12.2.13T, 12.2(8)YN	NA	Flexi6 support in Cisco IOS Release 12.2(8)YN. For extended EC configuration information for the Cisco 1700 series, see the following sections: • Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS7750 • Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS 7750 Example
Cisco 2600 series Cisco 2600XM Cisco 3600 series Cisco 3700 series Cisco VG200	NM-HDV (C549)	NA	12.2(13)T	NA	12.2(13)T	Full support
	NM-1V, (C542)	No	NA	No	NA	
Cisco AS5300		NA	No	NA	No	**Note** See your Cisco representative for information about EC support on the Cisco AS5300.
Cisco AS5350, Cisco AS5400, Cisco AS5850		NA	NA	NA	NA	Different DSP with its own 128ms-coverage EC
Cisco 7200 series	PA-VXx-2 TE1+ and PA-MCX-nTE1	NA	12.2(13)T	NA	12.2(13)T	PA-MCX-nTE1 port adapters do not have their own DSPs, so they use the DSPs of PA-VXx-2TE1+ port adapters. For extended EC configuration information for the Cisco 7200 series, see the "Changing Codec Complexity on the Cisco 7200 Series" section on page 17
Cisco 7500 series		NA	12.2(13)T	NA	No	No medium complexity

Table 1 Extended Echo Canceller Algorithm Coverage by Platform (continued)

Platform	Module	High Complexity		Medium Complexity		Comments
Cisco ICS7750		12.2(13)T, 12.2(13)ZH	12.2(13)T, 12.2(13)ZH	12.2(13)T, 12.2(8)YN	NA	Flexi6 support. For extended EC configuration information for the Cisco ICS7750, see the following sections: • Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS7750 • Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS 7750 Example
Cisco MC3810	HCM 549	12.2(13)T	12.2(13)T	NA	NA	Full support

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Restrictions

- Not all Cisco platforms that use C542 and C549 DSPs support the extended EC. All other platforms continue to use the Cisco-proprietary 32-ms EC by default.
- The G.168 extended EC is not supported on the Cisco AS5300 in this release. See your account representative for information about support for the extended EC on the Cisco AS5300.
- The Cisco 1700 series does not support the T1/E1 card in Cisco IOS Release 12.2(13)T.
- The NM-2V does not support the extended EC on the Cisco 2600, Cisco 2600XM, Cisco 3600 series, Cisco 3700 series, or Cisco VG200.

Information About Enhanced ITU-T G.168 Echo Cancellation

To configure the Enhanced ITU-T G.168 Echo Cancellation feature, you must understand the following concepts:

- Enhanced ITU-T G.168 Echo Cancellation, page 5

Enhanced ITU-T G.168 Echo Cancellation

Echo is the sound of your own voice reverberating in the telephone receiver while you are talking. When timed properly, echo is not a problem in the conversation; however, if the echo interval exceeds approximately 25 ms, it can be distracting to the speaker. Echo is controlled by ECs. By design, ECs are limited by the total amount of time they wait for the reflected speech to be received, which is known as an echo tail. The echo tail is normally 32 ms.

In the traditional telephony network, echo is generally caused by an impedance mismatch when the four-wire network is converted to the two-wire local loop. Echo cancellation is required because of packet network latency.

Echo cancellation is implemented in DSP firmware on the gateways and is independent of other functions implemented in the DSP (the DSP protocol and compression algorithm). In voice packet-based networks, ECs are built into the low-bit-rate codecs and are operated on each DSP. Figure 1 shows a common voice network where echo cancellation might be used, and Figure 2 shows a typical DSP channel configured for voice processing.

Figure 1 Echo Cancellation Network

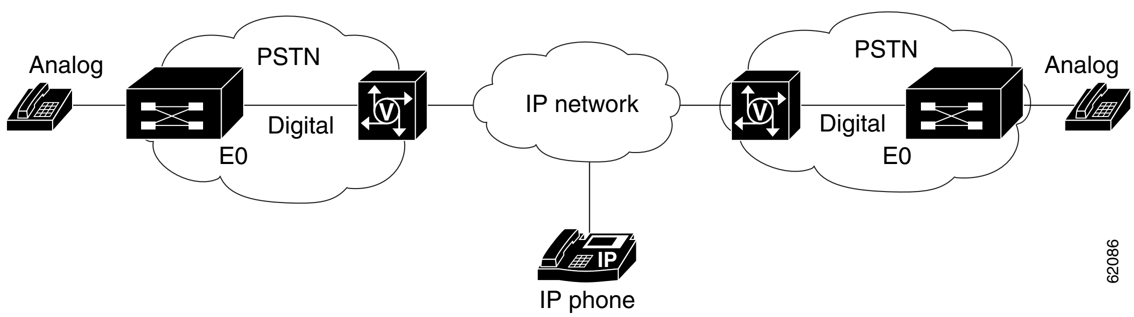
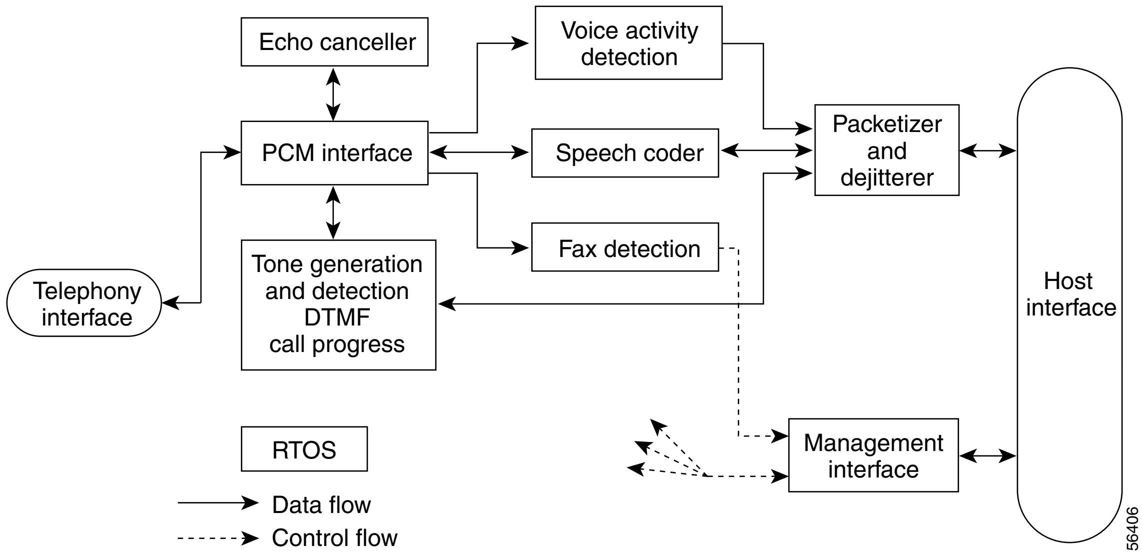


Figure 2 DSP Channel Configured for Voice Processing



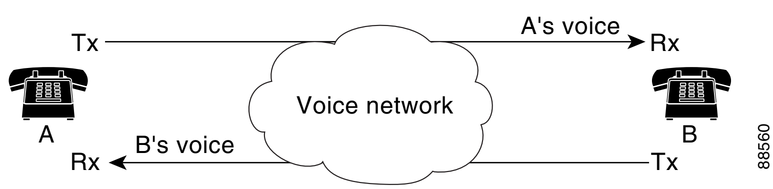
Voice Paths

Every voice conversation has at least two participants. From the perspective of each participant, there are two voice paths in every call:

- Transmit path (also called the send or Tx path)—The transmit path is created when a person speaks. The sound is transmitted from the mouth of the speaker to the ear of the listener.
- Receive path (also called the return or Rx path)—The receive path is created when a person hears the conversation. The sound is received by the ear of the listener from the mouth of the speaker.

Figure 3 shows a simple voice call between caller A and caller B. The top line represents the Tx path for caller A, which becomes the Rx path for caller B. The bottom line represents the Tx path for caller B, which becomes the Rx path for caller A.

Figure 3 Echo in a Voice Network



An echo canceller is a component of a voice gateway that reduces the level of echoes that leak from the Rx path (from the gateway out into the tail circuit) into the Tx path (from the tail circuit into the gateway). Rx and Tx here are from the perspective of the voice gateway.

Echo cancellers face into the PSTN tail circuit. They eliminate echoes in the tail circuit on its side of the network.

From the perspective of the echo canceller in a voice gateway, the Rx signal is a voice coming across the network from another location. The Tx signal is a mixture of the voice call in the other location and the echo of the original voice, which comes from the tail circuit on the initiating end and is sent to the receiving end.

The echo canceller in the originating gateway looks out into the tail circuit and is responsible for eliminating echo signal from the initiation Tx signal and allowing a voice call to go through unimpeded.

Note Delay and jitter in the WAN do not affect the operation of the echo canceller because the tail circuit, where the echo canceller operates, is static.

Basics of Echo Canceller Operation

An echo canceller removes the echo portion of the signal coming out of the tail circuit and headed into the WAN. It does so by learning the electrical characteristics of the tail circuit and forming its own model of the tail circuit in its memory, and creating an estimated echo signal based on the current and past Rx signal. It subtracts the estimated echo from the actual Tx signal coming out of the tail circuit. The quality of the estimation is continuously improved by monitoring the estimation error.

The analog circuit is known as the tail circuit. It forms the tail or termination of the call from the perspective of the person experiencing the echo.

A packet voice gateway is a gateway between a digital packet network and a public switched telephone network (PSTN). It can include both digital (TDM) and analog links.

The tail circuit is everything connected to the PSTN side of a packet voice gateway—all the switches, multiplexers, cabling, and PBXs between the voice gateway and the telephone.

Echo Canceller Components

A typical echo canceller includes two components: Convolution processor (CP) and a nonlinear processor (NLP).

Convolution processor

The CP first stage captures and stores the outgoing signal toward the far-end hybrid. The CP then switches to monitoring mode and, when the echo signal returns, estimates the level of the incoming echo signal and subtracts the attenuated original voice signal from the echo signal.

The time required to adjust the level of attenuation needed to the original signal is called the convergence time. Because the convergence process requires that the voice signal be stored in memory, the EC has limited coverage of tail circuit delay, normally 64 ms, 96 ms, and up to 128 ms. After convergence, the CP provides about 18 dB of echo return loss enhancement (ERLE). Because a typical analog phone circuit provides at least 12 dB of echo return loss (ERL) (that is, the echo path loss between the echo canceller and the far-end hybrid), the expected permanent ERL of the converged echo canceller is about 30 dB or greater.

Nonlinear processor

In single-talk mode, that is, when one person is talking and the other is silent, the NLP replaces the residual echo at the output of the echo canceller with comfort noise based on the actual background noise of the voice path. The background noise normally changes over the course of a phone conversation, so the NLP must adapt over time. The NLP provides an additional loss of at least 25 dB when activated. In double-talk mode, the NLP must be deactivated because it would create a one-way voice effect by adding 25 to 30 dB of loss in only one direction.

To completely eliminate the perception of echo, the Talker Echo Loudness Rating (TELR) should be greater than 65 dB in all situations. To reflect this reality, ITU-T Recommendation G.168 regarding echo canceller requires an ERL equal to or greater than 55 dB. Segmentation Local Reference (SLR), Receive Loudness Rating (RLR), and Cell Loss Ratio (CLR) along the echo path should allow another 10 dB to meet the expected TELR. CP, NLP and Loudness Ratings (LRs) must be optimized to make sure that echo is canceled effectively.

Echo

Following are descriptions of the primary measurements of relative signal levels used by echo cancellers. They are all expressed in dB.

- Echo return loss (ERL)—Reduction in the echo level produced by the tail circuit without the use of an echo canceller. If an Rx speech signal enters the tail circuit from the network at a level of X dB, the echo coming back from the tail circuit into the echo canceller is (X—ERL).
- Echo return loss enhancement (ERLE)—Additional reduction in echo level accomplished by the echo canceller. An echo canceller is not a perfect device; the best it can do is attenuate the level of the returning echo. ERLE is a measure of this echo attenuation. It is the difference between the echo level arriving from the tail circuit at the echo canceller and the level of the signal leaving the echo canceller.
- Acombined (ACOM)—Total ERL seen across the terminals of the echo canceller. ACOM is the sum of ERL + ERLE, or the total ERL seen by the network.

For more information about the echo canceller, refer to the *Echo Analysis for Voice over IP* document on Cisco.com.

Echo Canceller Coverage

Echo canceller coverage (also known as tail coverage or tail length) is the length of time that the echo canceller stores its approximation of an echo in memory. It is the maximum echo delay that an echo canceller is able to eliminate.

The echo canceller faces into a static tail circuit with input and an output. If a word enters a tail circuit, the echo is a series of delayed and attenuated versions of that word, depending on the number of echo sources and the delays associated with them. After a certain period of time, no more signal comes out. This time period is known as the ringing time of the tail circuit—the time required for all of the ripples to disperse. To fully eliminate all echoes, the coverage of the echo canceller must be as long as the ringing time of the tail circuit.

How to Configure Enhanced ITU-T G.168 Echo Cancellation

This section contains procedures for configuring the Enhanced ITU-T G.168 Echo Cancellation feature. Each procedure is identified as either required or optional.

- Switching Echo Cancellers, page 9 (optional)
- Configuring Echo Cancellation Parameters, page 18 (optional)
- Verifying Codec Complexity Settings, page 20 (optional)
- Verifying Analog and Digital Voice Port Configurations, page 21 (optional)

Switching Echo Cancellers

To add, switch, or remove the extended EC in high-complexity mode without reloading the router, perform the following tasks in the order listed:

1. Shutting Down All T1 Voice Ports
2. Shutting Down the T1 Controller
3. Changing Codec Complexity
4. Adding Back the PRI Groups and DS-0 Groups
5. Reapplying Voice-Port and Serial Interface Configurations
6. Reassigning Voice Ports to Dial-Peer Configurations
7. Bringing the T1 Controller Back Up

Note To switch ECs on the Cisco 1700 or Cisco ICS7750, you need use only the **codec complexity** command. See the "Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS7750" section on page 16 for configuration steps. See also the "Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS 7750 Example" section on page 58.

8. Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS7750

Shutting Down All T1 Voice Ports

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **voice-port** *slot*/*port*:*ds0-group*
4. **shutdown**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.
Step 2	**`configure`** `{`**`terminal`**`}` **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice-port** *slot***/***port***:***ds0-group* **Example:** `Router(config)# voice-port 1/0:23`	Enters voice port configuration mode on the selected slot, port, and DS-0 group.
Step 4	**shutdown** **Example:** `Router(config-voiceport)# shutdown`	Shuts down all voice ports assigned to the T1 interface on the voice card.
Step 5	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.

Shutting Down the T1 Controller

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **controller t1** *1/0*
4. **shutdown**
5. **no ds0-group** *ds0-group* **timeslots** *timeslot-list* **type** {**e&m-immediate** | **e&m-delay** | **e&m-wink** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

   Or
6. **no pri-group timeslots** *timeslot-list*
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.
Step 2	**configure** {**terminal**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**controller t1** *1/0* **Example:** `Router(config)# controller t1 1/0`	Enters controller configuration mode on the T1 controller on the selected slot and port.

Step 4	**`shutdown`** **Example:** `Router(config-controller)# shutdown`	Shuts down the T1 controller.
Step 5	**`no ds0-group`** *`ds0-group`* **`timeslots`** *`timeslot-list`* **`type`** {**`e&m-immediate`** | **`e&m-delay`** | **`e&m-wink`** | **`fxs-ground-start`** | **`fxs-loop-start`** | **`fxo-ground-start`** | **`fxo-loop-start`**} **Example:** `Router(config-controller)# ds0-group` *`ds0-group`* `timeslots` *`timeslot-list`* `type fxs-loop-start`	Defines the T1 or E1 channels for use by compressed voice calls and the signaling method that the router uses to connect to the private branch exchange (PBX) or central office (CO). **Note** If you are configuring PRI groups instead of DS0 groups, skip this step and proceed to Step 6.
Step 6	**`no pri-group timeslots`** *`timeslot-list`* **Example:** `Router(config-controller)# pri-group timeslots` *`timeslot-list`*	Specifies an ISDN Primary Rate Interface (PRI) on a channelized T1 or E1 controller. **Note** When configuring PRI groups, you must also configure the **isdn switch-type** command. Also, only one PRI group can be configured on a controller.
Step 7	**`exit`** **Example:** `Router(config-controller) exit`	Exits controller configuration mode on the T1 controller.

Changing Codec Complexity

Note If you are configuring a Cisco 7200 series, see the "Changing Codec Complexity on the Cisco 7200 Series" section on page 17 and also the "Changing Codec Complexity on the Cisco 7200 Series Example" section on page 58.

Note You must first clear all calls on the system before changing codec complexity.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **voice-card** *slot*
4. **codec complexity** {**high** | **medium**} [**ecan-extended**]
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure** {**terminal** | **memory** | **network**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-card** *slot* **Example:** `Router(config)# voice-card 1`	Enters voice-card configuration mode on the specified slot.

Step 4	**codec complexity** {**high** | **medium**} [**ecan-extended**]
	
	Example:
	`Router(voice-card)# codec complexity high ecan-extended`
	
	Example:
	`Router(voice-card)# codec complexity medium ecan-extended`
		Changes the codec complexity to high or medium and switches from the Cisco proprietary G.165-compliant EC (medium-complexity) to the extended EC (high-complexity).
		Note To switch echo cancellers on the Cisco 1700 series or Cisco ICS7750, see the "Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS7750" section on page 16 for configuration information.
		Note To change codec complexity on the Cisco 7200 series, see the "Changing Codec Complexity on the Cisco 7200 Series Example" section on page 58.

Step 5	**end** **Example:** `Router(voice-card)# end`	Exits voice-card configuration mode and completes the steps for changing the codec complexity and switching to the extended EC.

Adding Back the PRI Groups and DS-0 Groups

Note You must first clear all calls on the system before adding back PRI groups and DS-0 groups.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **controller t1** *1/0*
4. **ds0-group** *ds0-group* **timeslots** *timeslot-list* **type** {**e&m-immediate** | **e&m-delay** | **e&m-wink** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

Or

5. **pri-group timeslots** *timeslot-list*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure** {**terminal** | **memory** | **network**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**controller t1** *1/0* **Example:** `Router(config)# controller t1 1/0`	Enters controller configuration mode on the T1 controller on the selected slot and port.
Step 4	**ds0-group** *ds0-group* **timeslots** *timeslot-list* **type** {**e&m-immediate** | **e&m-delay** | **e&m-wink** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**} **Example:** Router(config-controller)# ds0-group *ds0-group* timeslots *timeslot-list* type fxs-loop-start	Defines the T1 or E1 channels for use by compressed voice calls and the signaling method that the router uses to connect to the private branch exchange (PBX) or central office (CO). **Note** If you are configuring PRI groups instead of DS0 groups, skip this step and proceed to Step 5.
Step 5	**pri-group timeslots** *timeslot-list* **Example:** Router(config-controller)# pri-group timeslots *timeslot-list*	Specifies an ISDN Primary Rate Interface (PRI) on a channelized T1 or E1 controller. **Note** When configuring PRI groups, you must also configure the **isdn switch-type** command. Also, only one PRI group can be configured on a controller.
Step 6	**exit** **Example:** `Router(config-controller)# exit`	Exits controller configuration mode and completes the process for adding back the PRI groups or DS-0 groups.

Reapplying Voice-Port and Serial Interface Configurations

Note You must first clear all calls on the system before reapplying voice-port and serial interface configurations.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **voice-port** *slot*/*port*:*ds0-group-no*
4. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure`** `{`**`terminal`** `|` **`memory`** `|` **`network`**`}` **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** *`slot`*`/`*`port`*`:`*`ds0-group-no`* **Example:** `Router(config)# voice-port 1/0:0`	Enters voice-port configuration mode and reapplies voice-port or serial interface configuration to the first T1 interface. • Each defined DS-0 group number is represented on a separate voice port. This allows you to define individual DS-0s on the digital T1/E1 card.
Step 4	**`exit`** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes reapplying voice-ports and serial interface configurations.

Reassigning Voice Ports to Dial-Peer Configurations

Note You must first clear all calls on the system before reassigning voice-ports to dial-peer configurations.

Note If you are using PRI, you might need to reapply the D channel configuration.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **dial-peer voice** *tag* **pots**
4. **port** *slot-number*/*subunit-number*/*port*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure** {**terminal**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **pots** **Example:** `Router(config)# dial-peer voice 133001 pots`	Enters dial-peer configuration mode and configures a POTS peer using a unique numeric identifier tag.
Step 4	**port** *slot-number*/*subunit-number*/*port* **Example:** `Router(config-dial-peer)# port 1/0/0`	Associates a dial peer with a specific voice port.
Step 5	**end** **Example:** `Router(config-dial-peer)# end`	Exits dial-peer configuration mode and completes the reassignment of voice ports to dial-peer configurations.

Bringing the T1 Controller Back Up

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **controller t1** *slot*/*port*
4. **no shutdown**
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.

Step 2	**configure** {**terminal**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**controller t1** *slot*/*port* **Example:** `Router(config)# controller t1 1/0`	Specifies the T1 controller slot and port and enters controller configuration mode.
Step 4	**no shutdown** **Example:** `Router(config-controller)# no shutdown`	Saves the controller configurations on the slot and port specified.
Step 5	**end** **Example:** `Router(config-controller)# end`	Exits controller configuration mode and completes the process to bring the T1 controller back up.

This completes the steps for switching ECs and configuring EC parameters on digital voice ports on the Cisco 2600 series, Cisco 2600XM, Cisco 3600 series, Cisco 3700 series, Cisco 7200 series, Cisco MC3810, and Cisco VG200.

Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS7750

The **codec complexity medium** command enables the extended echo canceller by default on the Cisco 1700 series and the Cisco ICS7750 in Cisco IOS release 12.2(13)ZH.

See Table 1 for extended EC algorithm coverage by platform.

Note You must clear all calls on the system before using the following commands. If there are active calls on the system, the commands are ignored and a warning message is issued.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **voice-card** *slot*
4. **codec complexity** {**high** | **medium**}
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.
Step 2	**configure** {**terminal**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-card** *slot* **Example:** `Router(config)# voice-card 1`	Enters voice card configuration mode on the specified slot.
Step 4	**codec complexity** {**medium**} **Example:** `Router(voice-card)# codec complexity medium`	Enables the extended EC (default).
Step 5	**end** **Example:** `Router(voice-card)# end`	Exits voice-card configuration mode and completes the steps for configuring the extended EC on the Cisco 1700 series and Cisco ICS7750.

Changing Codec Complexity on the Cisco 7200 Series

On the Cisco 7200 series, the PA-MCX-2TE1 port adapter (PA) card can be used for making voice calls. This PA does not have any DSPs but uses the DSP resources of the PA-VXC-2TE1+ card present in another slot. If the PA-MCX card is used, codec complexity is configured for PA-VXC, while all other echo cancellation configurations are done for PA-MCX.

The PA-MCX card borrows the DSP resources from the PA-VXC, PA-VXB, or PA-VXA cards. Even if one of the PA-VXC, PA-VXB, or PA-VXA cards has extended echo cancellation configured on the DSP interface, the extended echo cancellation CLI is enabled for the PA-MCX card. It is recommended that the same codec complexity and echo cancellation configurations be present on all the PA-VXC, PA-VXB, or PA-VXA cards in the router.

See Table 1 for extended EC algorithm coverage by platform.

Note You must clear all calls on the system before using the following commands. If there are active calls on the system, the commands are ignored and a warning message is issued.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}

3. **dspint dspfarm** *slot*/**0**
4. **codec complexity** {**high** | **medium**} [**ecan-extended**]
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.
Step 2	**configure** {**terminal**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dspint dspfarm** *slot*/**0** **Example:** `Router(config)# dspint dspfarm 2/0`	Enables the digital signal processor (DSP) interface on the specified slot and port.
Step 4	**codec complexity** {**high** | **medium**} [**ecan-extended**] **Example:** `Router(config-dspfarm)# codec complexity medium ecan-extended`	Changes the codec complexity to high or medium on the Cisco 7200 series.
Step 5	**end** **Example:** `Router(voice-card)# end`	Exits to global configuration mode and completes the steps for changing the codec complexity on the Cisco 7200 series.

Configuring Echo Cancellation Parameters

In Cisco voice implementations, ECs are enabled using the **echo cancel enable** command, and echo tails are configured using the **echo cancel coverage** command.

To configure parameters related to the extended EC, use the following commands beginning in user EXEC mode.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **voice-port** *slot/port*:*ds0-group-no*
4. **echo cancel enable**
5. **echo suppressor** *seconds*
6. **echo cancel coverage** {**8** | **16** | **24** | **32** | **48** | **64**}

7. **non-linear**
8. **echo cancel erl worst-case** {*6* | *3* | *0*}
9. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure** {**terminal**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *slot/port***:***ds0-group-no* **Example:** `Router(config)# voice-port 1/0:0`	Enters voice-port configuration mode on the selected slot, port, and DS-0 group. • Each defined DS-0 group number is represented on a separate voice port. This allows you to define individual DS-0s on the digital T1/E1 card.
Step 4	**echo cancel enable** **Example:** `Router(config-voiceport)# echo cancel enable`	Enables echo cancellation. • The Cisco G.165 EC is enabled by default with echo suppression off. The echo suppressor can be turned on only when the default Cisco G.165 EC is used. The **echo suppressor** command used with the default Cisco EC is still visible when the extended EC is selected, but it does not do anything. Use the **no** form of this command to disable the EC. **Note** For this command to work, the **echo cancel coverage** command must also be configured.
Step 5	**echo suppressor** *seconds* **Example:** `Router(config-voiceport)# echo suppressor`	(Optional) Applies echo suppression for the number of seconds specified. S • This command reduces the initial echo before the echo canceller can converge. In case of double-talk in the first number of seconds, the code automatically disables the suppressor. **Note** The echo canceller must be enabled for this command to work.

	Command or Action	Purpose
Step 6	**echo cancel coverage** {8 | 16 | 24 | 32 | 48 | 64} **Example:** `Router(config-voiceport)# echo cancel coverage 64`	Adjusts the size of the echo canceller (echo path capacity coverage). This command enables cancellation of voice that is sent out the interface and received back on the same interface within the configured amount of time. **Note** This command is valid only when the echo canceller feature has been enabled. See Step 4 above.
Step 7	**non-linear** **Example:** `Router(config-voiceport)# non-linear`	Selects nonlinear processing (residual echo suppression) in the EC, which either shuts off any signal or mixes in comfort noise if no near-end speech is detected. **Note** Echo cancelling must be enabled for this feature to work. See Step 4 above. • Nonlinear processing is enabled when the extended G.168 echo canceller is enabled. Use the **no** form of this command to disable the NLP. • The Cisco G.165 EC is enabled by default with the echo suppressor turned off. The echo suppressor can be turned on only when using the default Cisco G.165 EC is used. See the **echo suppressor** command. • The **echo suppressor** command used with the Cisco default EC is still visible when the extended EC is selected, but it does not do anything.
Step 8	**echo-cancel erl worst-case** [*0* | *3* | *6*] **Example:** `Router# echo-cancel erl worst-case 6`	Determines worst-case echo return loss (ERL) in decibels (dB). • This command is enabled by default with the G.168 extended EC.
Step 9	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes the configuration.

Verifying Codec Complexity Settings

To verify the codec complexity and extended EC configuration, enter the **show running-config** command to display the current voice-card setting. If medium complexity is specified, the codec complexity setting is not displayed. If high complexity is specified, the "codec complexity high" setting is displayed.

The following example shows abbreviated command output if high complexity is specified on the Cisco MC3810:

```
Router# show running-config
.
.
.
hostname router-alpha

voice-card 0
 codec complexity high ecan-extended
```

.
.
.

Verifying Analog and Digital Voice Port Configurations

After configuring the voice ports on your router, perform the following steps to verify proper operation.

Step 1 Pick up the handset of an attached telephony device and check for dial tone.

Step 2 If you have dial tone, check for dual-tone multifrequency (DTMF) detection. If dial tone stops when you dial a digit, the voice port most likely is configured properly.

Step 3 To identify port numbers of voice interfaces installed in your router, use the **show voice port summary** command. For examples of the output, refer to the "Show Voice Port Summary Samples" section of the *Configuring Voice Ports* document.

Step 4 To verify voice port parameter settings, enter the **show voice port** command. For sample output, refer to the "Show Voice Port Samples" section of the *Configuring Voice Ports* document.

Step 5 To display the active call information for voice calls or fax transmissions in progress, use the **show call active** command. This command displays information about call times, dial peers, connections, quality of service, and other status and statistical information. The **voice** keyword displays all voice calls currently connected through the router or access server.

Configuration Examples for Enhanced ITU-T G.168 Echo Cancellation

This section contains the following configuration examples:

- Enabling the Echo Canceller Example, page 21
- Switching the Echo Canceller Example, page 22
- Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS7750, page 16
- Changing Codec Complexity on the Cisco 7200 Series Example, page 58
- Adjusting the Echo Canceller Size Example, page 58
- Worst-Case Echo Return Loss Example, page 58
- Checking the Active Calls Example, page 59

Enabling the Echo Canceller Example

The following example enables extended echo cancellation and adjusts the size of the EC to 64 ms on a Cisco 3600 series router:

```
voice-port 1/0/0
 echo cancel enable
 echo cancel coverage 64
```

The following example enables extended echo cancellation and adjusts the size of the EC to 64 ms on a Cisco MC3810:

```
voice-port 1/1:0
 echo cancel enable
 echo cancel coverage 64
```

The following example enables the Enhanced ITU-T G.168 Echo Cancellation feature on a Cisco 1700 series or Cisco ICS 7750:

```
codec complexity medium
```

Switching the Echo Canceller Example

The following examples show that the default Cisco-proprietary EC has been switched to the extended EC. These examples show voice and POTS dial peers on originating and terminating router pairs running the maximum number of calls (23) on a single T1 interface.

The following is example **show running config** output from an originating Cisco 3640:

```
Router# show running config
!
version 12.2
service timestamps debug datetime msec
service timestamps log uptime
no service password-encryption
!
hostname 3640echo-135-hc
!
enable password xxx
!
voice-card 1
 codec complexity high ecan-extended
!
ip subnet-zero
!
ip domain-name cisco.com
ip host santa 172.16.1.0
ip name-server 172.16.0.0
!
frame-relay switching
isdn switch-type primary-5ess
isdn voice-call-failure 0
call rsvp-sync
!
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
controller T1 1/1
 framing sf
 linecode ami
!
interface Ethernet0/0
 ip address 172.16.0.1 255.0.0.0
 half-duplex
!
interface Serial0/0
 no ip address
 encapsulation frame-relay
 no ip route-cache
```

```
 no ip mroute-cache
 no keepalive
 no fair-queue
 clock rate 256000
 no arp frame-relay
 cdp enable
 frame-relay traffic-shaping
 frame-relay interface-dlci 100
  class fr200
  vofr cisco
 hold-queue 1024 out
!
interface Ethernet0/1
 ip address 10.1.0.103 255.0.0.0
 full-duplex
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
!
ip classless
ip route 10.2.0.0 255.0.0.0 e0/1
ip route 172.16.0.0 255.0.0.0 172.16.0.1
ip route 172.16.0.1.0 255.0.0.0 172.17.0.0
no ip http server
ip pim bidir-enable
!
map-class frame-relay fr200
 frame-relay traffic-rate 560000 560000
 no frame-relay adaptive-shaping
 frame-relay cir 100000
 frame-relay mincir 100000
 frame-relay fair-queue
 frame-relay voice bandwidth 560000
!
voice-port 1/0:23
!
voice-port 2/0/0
!
voice-port 2/0/1
!
voice-port 2/1/0
!
voice-port 2/1/1
!
voice-port 3/0/0
!
voice-port 3/0/1
!
dial-peer cor custom
!
dial-peer voice 104001 voip
 destination-pattern 104001
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g711alaw
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 104002 voip
 destination-pattern 104002
```

```
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g711ulaw
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 104003 voip
 destination-pattern 104003
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g726r16
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 104004 voip
 destination-pattern 104004
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g726r24
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 104005 voip
 destination-pattern 104005
 session target ipv4:10.2.0.104
 dtmf-relay h245-alphanumeric
 codec g726r32
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 104006 voip
 destination-pattern 104006
 session target ipv4:10.2.0.104
 dtmf-relay h245-alphanumeric
 codec g728
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 104007 voip
 destination-pattern 104007
 session target ipv4:10.2.0.104
 dtmf-relay h245-alphanumeric
 codec g729br8
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104008 voip
 destination-pattern 104008
 session target ipv4:10.2.0.104
 dtmf-relay h245-alphanumeric
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104009 voip
 destination-pattern 104009
 session target ipv4:10.2.0.104
 dtmf-relay h245-signal
 codec gsmefr
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104010 voip
 destination-pattern 104010
 session target ipv4:10.2.0.104
```

```
 dtmf-relay h245-signal
 codec gsmfr
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104011 voip
 destination-pattern 104011
 session target ipv4:10.2.0.104
 dtmf-relay h245-signal
 codec g723r53
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104012 voip
 destination-pattern 104012
 session target ipv4:10.2.0.104
 dtmf-relay h245-signal
 codec g723r63
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104101 vofr
 destination-pattern 104101
 session target Serial0/0 100
 dtmf-relay
 codec g726r16
 fax rate 14400
!
dial-peer voice 104102 vofr
 destination-pattern 104102
 session target Serial0/0 100
 dtmf-relay
 codec g726r24
 fax rate 14400
!
dial-peer voice 104103 vofr
 destination-pattern 104103
 session target Serial0/0 100
 dtmf-relay
 codec g726r32
 fax rate 14400
!
dial-peer voice 104104 vofr
 destination-pattern 104104
 session target Serial0/0 100
 dtmf-relay
 codec g728
 fax rate 14400
!
dial-peer voice 104105 vofr
 destination-pattern 104105
 session target Serial0/0 100
 dtmf-relay
 codec g729br8
 fax rate 14400
!
dial-peer voice 104106 vofr
 destination-pattern 104106
 session target Serial0/0 100
 dtmf-relay
 fax rate 14400
!
dial-peer voice 104107 vofr
 destination-pattern 104107
```

```
 session target Serial0/0 100
 dtmf-relay
 codec g723r53
 fax rate 14400
!
dial-peer voice 104108 vofr
 destination-pattern 104108
 session target Serial0/0 100
 dtmf-relay
 codec g723r63
 fax rate 14400
!
dial-peer voice 104109 vofr
 destination-pattern 104109
 session target Serial0/0 100
 dtmf-relay
 codec g723ar53
 fax rate 14400
!
dial-peer voice 104110 vofr
 destination-pattern 104110
 session target Serial0/0 100
 dtmf-relay
 codec g723ar63
 fax rate 14400
!
dial-peer voice 104111 vofr
 destination-pattern 104111
 session target Serial0/0 100
 dtmf-relay
 codec g711alaw
 fax rate 14400
!
dial-peer voice 104112 vofr
 destination-pattern 104112
 session target Serial0/0 100
 dtmf-relay
 codec g711ulaw
 fax rate 14400
!
dial-peer voice 2001 pots
 incoming called-number 5440001
 port 1/0:23
!
dial-peer voice 2002 pots
 incoming called-number 5440002
 port 1/0:23
!
dial-peer voice 2003 pots
 incoming called-number 5440003
 port 1/0:23
!
dial-peer voice 2004 pots
 incoming called-number 5440004
 port 1/0:23
!
dial-peer voice 2005 pots
 incoming called-number 5440005
 port 1/0:23
!
dial-peer voice 2006 pots
 incoming called-number 5440006
 port 1/0:23
!
```

```
dial-peer voice 2007 pots
 incoming called-number 5440007
 port 1/0:23
!
dial-peer voice 2008 pots
 incoming called-number 5440008
 port 1/0:23
!
dial-peer voice 2009 pots
 incoming called-number 5440009
 port 1/0:23
!
dial-peer voice 2010 pots
 incoming called-number 5440010
 port 1/0:23
!
dial-peer voice 2011 pots
 incoming called-number 5440011
 port 1/0:23
!
dial-peer voice 2012 pots
 incoming called-number 5440012
 port 1/0:23
!
dial-peer voice 2013 pots
 incoming called-number 5440013
 port 1/0:23
!
dial-peer voice 2014 pots
 incoming called-number 5440014
 port 1/0:23
!
dial-peer voice 2015 pots
 incoming called-number 5440015
 port 1/0:23
!
dial-peer voice 2016 pots
 incoming called-number 5440016
 port 1/0:23
!
dial-peer voice 2017 pots
 incoming called-number 5440017
 port 1/0:23
!
dial-peer voice 2018 pots
 incoming called-number 5440018
 port 1/0:23
!
dial-peer voice 2019 pots
 incoming called-number 5440019
 port 1/0:23
!
dial-peer voice 2020 pots
 incoming called-number 5440020
 port 1/0:23
!
dial-peer voice 2021 pots
 incoming called-number 5440021
 port 1/0:23
!
dial-peer voice 2022 pots
 incoming called-number 5440022
 port 1/0:23
!
```

```
dial-peer voice 2023 pots
 incoming called-number 5440023
 port 1/0:23
!
dial-peer voice 2024 pots
 incoming called-number 5440024
 port 1/0:23
!
dial-peer voice 104301 voip
 destination-pattern 5481320
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g711alaw
 fax rate 14400
!
dial-peer voice 104302 voip
 destination-pattern 5481321
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g711ulaw
 fax rate 14400
!
dial-peer voice 104303 voip
 destination-pattern 5481322
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g726r16
 fax rate 14400
!
dial-peer voice 104304 voip
 destination-pattern 5481323
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g726r24
 fax rate 14400
!
dial-peer voice 104305 voip
 destination-pattern 5481324
 session target ipv4:10.2.0.104
 dtmf-relay h245-alphanumeric
 codec g726r32
 fax rate 14400
!
dial-peer voice 104306 voip
 destination-pattern 5481325
 session target ipv4:10.2.0.104
 dtmf-relay h245-alphanumeric
 codec g728
 fax rate 14400
!
dial-peer voice 104307 voip
 destination-pattern 5481326
 session target ipv4:10.2.0.104
 dtmf-relay h245-alphanumeric
 codec g729br8
 fax rate 14400
!
dial-peer voice 104401 voip
 destination-pattern 5481420
 session target ipv4:10.2.0.104
 dtmf-relay h245-alphanumeric
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
```

```
dial-peer voice 104402 voip
 destination-pattern 5481421
 session target ipv4:10.2.0.104
 dtmf-relay h245-signal
 codec gsmefr
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104403 voip
 destination-pattern 5481422
 session target ipv4:10.2.0.104
 dtmf-relay h245-signal
 codec gsmfr
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104404 voip
 destination-pattern 5481423
 session target ipv4:10.2.0.104
 dtmf-relay h245-signal
 codec g723r53
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104405 voip
 destination-pattern 5481424
 session target ipv4:10.2.0.104
 dtmf-relay h245-signal
 codec g723r63
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104406 voip
 destination-pattern 5481425
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g723ar53
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104407 voip
 destination-pattern 5481426
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g723ar63
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 104308 vofr
 destination-pattern 5481327
 session target Serial0/0 100
 dtmf-relay
 codec g711alaw
 fax rate 14400
!
dial-peer voice 104309 vofr
 destination-pattern 5481328
 session target Serial0/0 100
 dtmf-relay
 codec g711ulaw
fax rate 14400
!
dial-peer voice 104310 vofr
 destination-pattern 5481329
```

```
 session target Serial0/0 100
 dtmf-relay
 codec g726r16
 fax rate 14400
!
dial-peer voice 104311 vofr
 destination-pattern 5481330
 session target Serial0/0 100
 dtmf-relay
 codec g726r24
 fax rate 14400
!
dial-peer voice 104312 vofr
 destination-pattern 5481331
 session target Serial0/0 100
 dtmf-relay
 codec g726r32
 fax rate 14400
!
dial-peer voice 104313 vofr
 destination-pattern 5481332
 session target Serial0/0 100
 dtmf-relay
 codec g728
 fax rate 14400
!
dial-peer voice 104314 vofr
 destination-pattern 5481333
 session target Serial0/0 100
 dtmf-relay
 codec g729br8
 fax rate 14400
!
dial-peer voice 104408 vofr
 destination-pattern 5481427
 session target Serial0/0 100
 dtmf-relay
 fax rate 14400
!
dial-peer voice 104409 vofr
 destination-pattern 5481428
 session target Serial0/0 100
 dtmf-relay
 codec g729br8
 fax rate 14400
!
dial-peer voice 104410 vofr
 destination-pattern 5481429
 session target Serial0/0 100
 dtmf-relay
 fax rate 14400
!
dial-peer voice 104411 vofr
 destination-pattern 5481430
 session target Serial0/0 100
 dtmf-relay
 codec g723r53
 fax rate 14400
!
dial-peer voice 104412 vofr
 destination-pattern 5481431
 session target Serial0/0 100
 dtmf-relay
 codec g723r63
```

```
 fax rate 14400
!
dial-peer voice 104413 vofr
 destination-pattern 5481432
 session target Serial0/0 100
 dtmf-relay
 codec g723ar53
 fax rate 14400
!
dial-peer voice 104414 vofr
 destination-pattern 5481433
 session target Serial0/0 100
 dtmf-relay
 codec g723ar63
 fax rate 14400
!
dial-peer voice 135300 pots
 incoming called-number 54813..
 destination-pattern 135300
 port 3/0/0
!
dial-peer voice 135301 pots
 incoming called-number 54814..
 destination-pattern 135301
 port 3/0/1
!
line con 0
 exec-timeout 0 0
 timeout login response 0
line aux 0
line vty 0 4
 exec-timeout 0 0
 password lab
 login
!
end
```

The following is example **show running config** output from a terminating Cisco 3640:

```
version 12.2
service timestamps debug datetime msec
service timestamps log uptime
no service password-encryption
!
hostname 3640eb-139-hc
!
enable password lab
!
voice-card 1
 codec complexity high ecan-extended
!
ip subnet-zero
!
ip domain-name cisco.com
ip host santa 172.16.0.0
ip name-server 172.16.0.0
!
isdn switch-type primary-5ess
isdn voice-call-failure 0
call rsvp-sync
!
controller T1 1/0
 framing esf
 linecode b8zs
```

```
 pri-group timeslots 1-24
!
controller T1 1/1
 framing sf
 linecode ami
!
interface Ethernet0/0
 ip address 172.16.0.0 255.0.0.0
 half-duplex
!
interface Serial0/0
 bandwidth 2000000
 no ip address
 encapsulation frame-relay
 no ip route-cache
 no ip mroute-cache
 no keepalive
 no fair-queue
 frame-relay traffic-shaping
 frame-relay interface-dlci 100
  class fr200
  vofr cisco
 hold-queue 1024 out
!
interface Ethernet0/1
 ip address 10.2.0.104 255.0.0.0
 full-duplex
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice modem
!
ip classless
ip route 10.1.0.0 255.0.0.0 e0/1
ip route 172.16.0.1 255.o.0.0 172.16.0.0
ip route 172.17.0.0.0 255.o.255.0 172.16.0.0
no ip http server
ip pim bidir-enable
!
map-class frame-relay fr200
 frame-relay traffic-rate 560000 560000
 no frame-relay adaptive-shaping
 frame-relay cir 100000
 frame-relay mincir 100000
 frame-relay fair-queue
 frame-relay voice bandwidth 560000
!
voice-port 1/0:23
!
voice-port 2/1/0
!
voice-port 2/1/1
!
voice-port 3/0/0
!
voice-port 3/0/1
!
voice-port 3/1/0
!
voice-port 3/1/1
!
dial-peer cor custom
```

```
!
!
!
dial-peer voice 104001 pots
 destination-pattern 104001
 port 1/0:23
 prefix 5500001
!
dial-peer voice 104002 pots
 destination-pattern 104002
 port 1/0:23
 prefix 5500002
!
dial-peer voice 104003 pots
 destination-pattern 104003
 port 1/0:23
 prefix 5500003
!
dial-peer voice 104004 pots
 destination-pattern 104004
 port 1/0:23
 prefix 5500004
!
dial-peer voice 104005 pots
 destination-pattern 104005
 port 1/0:23
 prefix 5500005
!
dial-peer voice 104006 pots
 destination-pattern 104006
 port 1/0:23
 prefix 5500006
!
dial-peer voice 104007 pots
 destination-pattern 104007
 port 1/0:23
 prefix 5500007
!
dial-peer voice 104008 pots
 destination-pattern 104008
 port 1/0:23
 prefix 5500008
!
dial-peer voice 104009 pots
 destination-pattern 104009
 port 1/0:23
 prefix 5500009
!
dial-peer voice 104010 pots
 destination-pattern 104010
 port 1/0:23
 prefix 5500010
!
dial-peer voice 104011 pots
 destination-pattern 104011
 port 1/0:23
 prefix 5500011
!
dial-peer voice 104012 pots
 destination-pattern 104012
 port 1/0:23
 prefix 5500012
!
dial-peer voice 104101 pots
```

```
 destination-pattern 104101
 port 1/0:23
 prefix 5500013
!
dial-peer voice 104102 pots
 destination-pattern 104102
 port 1/0:23
 prefix 5500014
!
dial-peer voice 104103 pots
 destination-pattern 104103
 port 1/0:23
 prefix 5500015
!
dial-peer voice 104104 pots
 destination-pattern 104104
 port 1/0:23
 prefix 5500016
!
dial-peer voice 104105 pots
 destination-pattern 104105
 port 1/0:23
 prefix 5500017
!
dial-peer voice 104106 pots
 destination-pattern 104106
 port 1/0:23
 prefix 5500018
!
dial-peer voice 104107 pots
 destination-pattern 104107
 port 1/0:23
 prefix 5500019
!
dial-peer voice 104108 pots
 destination-pattern 104108
 port 1/0:23
 prefix 5500020
!
dial-peer voice 104109 pots
 destination-pattern 104109
 port 1/0:23
 prefix 5500021
!
dial-peer voice 104110 pots
 destination-pattern 104110
 port 1/0:23
 prefix 5500022
!
dial-peer voice 104111 pots
 destination-pattern 104111
 port 1/0:23
 prefix 5500023
!
dial-peer voice 104112 pots
 destination-pattern 104112
 port 1/0:23
 prefix 5500024
!
dial-peer voice 104301 pots
 destination-pattern 5481320
 port 2/1/0
 prefix ,,5500001
!
```

```
dial-peer voice 104310 pots
 destination-pattern 5481329
 port 2/1/0
 prefix ,,5500005
!
dial-peer voice 104311 pots
 destination-pattern 5481330
 port 2/1/0
 prefix ,,5500007
!
dial-peer voice 103001 voip
 incoming called-number 104001
 destination-pattern 103001
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g711alaw
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 103002 voip
 incoming called-number 104002
 destination-pattern 103002
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g711ulaw
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 103003 voip
 incoming called-number 104003
 destination-pattern 103003
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g726r16
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 103004 voip
 incoming called-number 104004
 destination-pattern 103004
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g726r24
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 103005 voip
 incoming called-number 104005
 destination-pattern 103005
 session target ipv4:10.1.0.103
 dtmf-relay h245-alphanumeric
 codec g726r32
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 103006 voip
 incoming called-number 104006
 destination-pattern 103006
 session target ipv4:10.1.0.103
 dtmf-relay h245-alphanumeric
 codec g728
 fax rate 14400
 fax protocol cisco
!
```

```
dial-peer voice 103007 voip
 incoming called-number 104007
 destination-pattern 103007
 session target ipv4:10.1.0.103
 dtmf-relay h245-alphanumeric
 codec g729br8
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103008 voip
 incoming called-number 104008
 destination-pattern 103008
 session target ipv4:10.1.0.103
 dtmf-relay h245-alphanumeric
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103009 voip
 incoming called-number 104009
 destination-pattern 103009
 session target ipv4:10.1.0.103
 dtmf-relay h245-signal
 codec gsmefr
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103010 voip
 incoming called-number 104010
 destination-pattern 103010
 session target ipv4:10.1.0.103
 dtmf-relay h245-signal
 codec gsmfr
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103011 voip
 incoming called-number 104011
 destination-pattern 103011
 session target ipv4:10.1.0.103
 dtmf-relay h245-signal
 codec g723r53
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103012 voip
 incoming called-number 104012
 destination-pattern 103012
 session target ipv4:10.1.0.103
 dtmf-relay h245-signal
 codec g723r63
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103101 vofr
 incoming called-number 104101
 destination-pattern 103101
 session target Serial0/0 100
 dtmf-relay
 codec g726r16
 fax rate 14400
!
dial-peer voice 103102 vofr
 incoming called-number 104102
 destination-pattern 103102
```

```
 session target Serial0/0 100
 dtmf-relay
 codec g726r24
 fax rate 14400
!
dial-peer voice 103103 vofr
 incoming called-number 104103
 destination-pattern 103103
 session target Serial0/0 100
 dtmf-relay
 codec g726r32
 fax rate 14400
!
dial-peer voice 103104 vofr
 incoming called-number 104104
 destination-pattern 103104
 session target Serial0/0 100
 dtmf-relay
 codec g728
 fax rate 14400
!
dial-peer voice 103105 vofr
 incoming called-number 104105
 destination-pattern 103105
 session target Serial0/0 100
 dtmf-relay
 codec g729br8
 fax rate 14400
!
dial-peer voice 103106 vofr
 incoming called-number 104106
 destination-pattern 103106
 session target Serial0/0 100
 dtmf-relay
 fax rate 14400
!
dial-peer voice 103107 vofr
 incoming called-number 104107
 destination-pattern 103107
 session target Serial0/0 100
 dtmf-relay
 codec g723r53
 fax rate 14400
!
dial-peer voice 103108 vofr
 incoming called-number 104108
 destination-pattern 103108
 session target Serial0/0 100
 dtmf-relay
 codec g723r63
 fax rate 14400
!
dial-peer voice 103109 vofr
 incoming called-number 104109
 destination-pattern 103109
 session target Serial0/0 100
 dtmf-relay
 codec g723ar53
 fax rate 14400
!
dial-peer voice 103110 vofr
 incoming called-number 104110
 destination-pattern 103110
 session target Serial0/0 100
```

```
 dtmf-relay
 codec g723ar63
 fax rate 14400
!
dial-peer voice 103111 vofr
 incoming called-number 104111
 destination-pattern 103111
 session target Serial0/0 100
 dtmf-relay
 codec g711alaw
 fax rate 14400
!
dial-peer voice 103112 vofr
 incoming called-number 104112
 destination-pattern 103112
 session target Serial0/0 100
 dtmf-relay
 codec g711ulaw
 fax rate 14400
!
dial-peer voice 104302 pots
 destination-pattern 5481321
 port 2/1/0
 prefix ,,5500003
!
dial-peer voice 104303 pots
 destination-pattern 5481322
 port 2/1/0
 prefix ,,5500005
!
dial-peer voice 104304 pots
 destination-pattern 5481323
 port 2/1/0
 prefix ,,5500007
!
dial-peer voice 104305 pots
 destination-pattern 5481324
 port 2/1/0
 prefix ,,5500009
!
dial-peer voice 104306 pots
 destination-pattern 5481325
 port 2/1/0
 prefix ,,5500011
!
dial-peer voice 104307 pots
 destination-pattern 5481326
 port 2/1/0
 prefix ,,5500013
!
dial-peer voice 104401 pots
 destination-pattern 5481420
 port 2/1/1
 prefix ,,5500002
!
dial-peer voice 104402 pots
 destination-pattern 5481421
 port 2/1/1
 prefix ,,5500004
!
dial-peer voice 104403 pots
 destination-pattern 5481422
 port 2/1/1
 prefix ,,5500006
```

```
!
dial-peer voice 104404 pots
 destination-pattern 5481423
 port 2/1/1
 prefix ,,5500008
!
dial-peer voice 104405 pots
 destination-pattern 5481424
 port 2/1/1
 prefix ,,5500010
!
dial-peer voice 104406 pots
 destination-pattern 5481425
 port 2/1/1
 prefix ,,5500012
!
dial-peer voice 104407 pots
 destination-pattern 5481426
 port 2/1/1
 prefix ,,5500014
!
dial-peer voice 104308 pots
 destination-pattern 5481327
 port 2/1/0
 prefix ,,5500001
!
dial-peer voice 104309 pots
 destination-pattern 5481328
 port 2/1/0
 prefix ,,5500003
!
dial-peer voice 104312 pots
 destination-pattern 5481331
 port 2/1/0
 prefix ,,5500009
!
dial-peer voice 104313 pots
 destination-pattern 5481332
 port 2/1/0
 prefix ,,5500011
!
dial-peer voice 104314 pots
 destination-pattern 5481333
 port 2/1/0
 prefix ,,5500013
!
dial-peer voice 104408 pots
 destination-pattern 5481427
 port 2/1/1
 prefix ,,5500002
!
dial-peer voice 104409 pots
 destination-pattern 5481428
 port 2/1/1
 prefix ,,5500004
!
dial-peer voice 104410 pots
 destination-pattern 5481429
 port 2/1/1
 prefix ,,5500006
!
dial-peer voice 104411 pots
 destination-pattern 5481430
 port 2/1/1
```

```
 prefix ,,5500008
!
dial-peer voice 104412 pots
 destination-pattern 5481431
 port 2/1/1
 prefix ,,5500010
!
dial-peer voice 104413 pots
 destination-pattern 5481432
 port 2/1/1
 prefix ,,5500012
!
dial-peer voice 104414 pots
 destination-pattern 5481433
 port 2/1/1
 prefix ,,5500014
!
dial-peer voice 103301 voip
 incoming called-number 5481320
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g711alaw
 fax rate 14400
!
dial-peer voice 103302 voip
 incoming called-number 5481321
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g711ulaw
 fax rate 14400
!
dial-peer voice 103303 voip
 incoming called-number 5481322
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g726r16
 fax rate 14400
!
dial-peer voice 103304 voip
 incoming called-number 5481323
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g726r24
 fax rate 14400
!
dial-peer voice 103305 voip
 incoming called-number 5481324
 session target ipv4:10.1.0.103
 dtmf-relay h245-alphanumeric
 codec g726r32
 fax rate 14400
!
dial-peer voice 103306 voip
 incoming called-number 5481325
 session target ipv4:10.1.0.103
 dtmf-relay h245-alphanumeric
 codec g728
 fax rate 14400
!
dial-peer voice 103307 voip
 incoming called-number 5481326
 session target ipv4:10.1.0.103
 dtmf-relay h245-alphanumeric
 codec g729br8
```

```
 fax rate 14400
!
dial-peer voice 103401 voip
 incoming called-number 5481420
 session target ipv4:10.1.0.103
 dtmf-relay h245-alphanumeric
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103402 voip
 incoming called-number 5481421
 session target ipv4:10.1.0.103
 dtmf-relay h245-signal
 codec gsmefr
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103403 voip
 incoming called-number 5481422
 session target ipv4:10.1.0.103
 dtmf-relay h245-signal
 codec gsmfr
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103404 voip
 incoming called-number 5481423
 session target ipv4:10.1.0.103
 dtmf-relay h245-signal
 codec g723r53
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103405 voip
 incoming called-number 5481424
 session target ipv4:10.1.0.103
 dtmf-relay h245-signal
 codec g723r63
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103406 voip
 incoming called-number 5481425
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g723ar53
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103407 voip
 incoming called-number 5481426
 session target ipv4:10.1.0.103
 dtmf-relay cisco-rtp
 codec g723ar63
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 103308 vofr
 incoming called-number 5481327
 session target Serial0/0 100
 dtmf-relay
 codec g711alaw
 fax rate 14400
!
```

```
dial-peer voice 103309 vofr
 incoming called-number 5481328
 session target Serial0/0 100
 dtmf-relay
 codec g711ulaw
 fax rate 14400
!
dial-peer voice 103310 vofr
 incoming called-number 5481329
 session target Serial0/0 100
 dtmf-relay
 codec g726r16
 fax rate 14400
!
dial-peer voice 103311 vofr
 incoming called-number 5481330
 session target Serial0/0 100
 dtmf-relay
 codec g726r24
 fax rate 14400
!
dial-peer voice 103312 vofr
 incoming called-number 5481331
 session target Serial0/0 100
 dtmf-relay
 codec g726r32
 fax rate 14400
!
dial-peer voice 103313 vofr
 incoming called-number 5481332
 session target Serial0/0 100
 dtmf-relay
 codec g728
 fax rate 14400
!
dial-peer voice 103314 vofr
 incoming called-number 5481333
 session target Serial0/0 100
 dtmf-relay
 codec g729br8
 fax rate 14400
!
dial-peer voice 103408 vofr
 incoming called-number 5481427
 session target Serial0/0 100
 dtmf-relay
 fax rate 14400
!
dial-peer voice 103409 vofr
 incoming called-number 5481428
 session target Serial0/0 100
 dtmf-relay
 fax rate 14400
!
dial-peer voice 103410 vofr
 incoming called-number 5481429
 session target Serial0/0 100
 dtmf-relay
 fax rate 14400
!
dial-peer voice 103411 vofr
 incoming called-number 5481430
 session target Serial0/0 100
 dtmf-relay
```

```
 codec g723r53
 fax rate 14400
!
dial-peer voice 103412 vofr
 incoming called-number 5481431
 session target Serial0/0 100
 dtmf-relay
 codec g723r63
 fax rate 14400
!
dial-peer voice 103413 vofr
 incoming called-number 5481432
 session target Serial0/0 100
 dtmf-relay
 codec g723ar53
 fax rate 14400
!
dial-peer voice 103414 vofr
 incoming called-number 5481433
 session target Serial0/0 100
 dtmf-relay
 codec g723ar63
 fax rate 14400
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 password lab
 login
!
end
```

The following is example **show running config** output from the originating Cisco MC3810:

```
Router# show running config
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 3810eb-18-hc-atm
!
enable password lab
!
network-clock base-rate 56k
ip subnet-zero
ip domain-name cisco.com
ip host santa 172.16.0.0
ip name-server 172.16.0.0
ip name-server 172.16.0.0
!
frame-relay switching
isdn voice-call-failure 0
call rsvp-sync
!
voice service voatm
 !
 session protocol aal2
  cac master
!
no voice confirmation-tone
voice-card 0
```

```
 codec complexity high ecan-extended
!
controller T1 0
 mode atm
 framing esf
 clock source inter
 linecode b8zs
!
controller T1 1
 mode cas
 framing esf
 linecode b8zs
 ds0-group 0 timeslots 1 type e&m-wink-start
 ds0-group 1 timeslots 2 type e&m-wink-start
 ds0-group 2 timeslots 3 type e&m-wink-start
 ds0-group 3 timeslots 4 type e&m-wink-start
 ds0-group 4 timeslots 5 type e&m-wink-start
 ds0-group 5 timeslots 6 type e&m-wink-start
 ds0-group 6 timeslots 7 type e&m-wink-start
 ds0-group 7 timeslots 8 type e&m-wink-start
 ds0-group 8 timeslots 9 type e&m-wink-start
 ds0-group 9 timeslots 10 type e&m-wink-start
 ds0-group 10 timeslots 11 type e&m-wink-start
 ds0-group 11 timeslots 12 type e&m-wink-start
 ds0-group 12 timeslots 13 type e&m-wink-start
 ds0-group 13 timeslots 14 type e&m-wink-start
 ds0-group 14 timeslots 15 type e&m-wink-start
 ds0-group 15 timeslots 16 type e&m-wink-start
 ds0-group 16 timeslots 17 type e&m-wink-start
 ds0-group 17 timeslots 18 type e&m-wink-start
 ds0-group 18 timeslots 19 type e&m-wink-start
 ds0-group 19 timeslots 20 type e&m-wink-start
 ds0-group 20 timeslots 21 type e&m-wink-start
 ds0-group 21 timeslots 22 type e&m-wink-start
 ds0-group 22 timeslots 23 type e&m-wink-start
 ds0-group 23 timeslots 24 type e&m-wink-start
!
process-max-time 100
!
interface Ethernet0
 ip address 172.16.0.0 255.0.0.0
 no ip route-cache
 no ip mroute-cache
!
interface Serial0
 no ip address
 encapsulation frame-relay
 no ip route-cache
 no ip mroute-cache
 no keepalive
 no fair-queue
 clockrate 250000
 no arp frame-relay
 cdp enable
 frame-relay traffic-shaping
 frame-relay interface-dlci 100
  class fr200
  vofr cisco
 frame-relay intf-type dce
 hold-queue 1024 out
!
interface Serial1
 no ip address
 no ip route-cache
```

```
 no ip mroute-cache
 shutdown
 no cdp enable
!
interface ATM0
 no ip address
 no atm ilmi-keepalive
 pvc 1/100
  vbr-rt 1536 1536 65535
  encapsulation aal2
 !
interface FR-ATM20
 no ip address
 no ip route-cache
 shutdown
!
ip classless
ip route 0.0.0.0 0.0.0.0 172.16.0.1
no ip http server
!
!
map-class frame-relay fr200
 frame-relay voice bandwidth 560000
 frame-relay traffic-rate 560000 560000
 no frame-relay adaptive-shaping
 frame-relay cir 100000
 frame-relay mincir 100000
 frame-relay fair-queue
!
voice-port 1:0
!
voice-port 1:1
!
voice-port 1:2
!
voice-port 1:3
!
voice-port 1:4
!
voice-port 1:5
!
voice-port 1:6
!
voice-port 1:7
!
voice-port 1:8
!
voice-port 1:9
!
voice-port 1:10
!
voice-port 1:11
!
voice-port 1:12
!
voice-port 1:13
!
voice-port 1:14
!
voice-port 1:15
!
voice-port 1:16
 timeouts wait-release 3
 connection trunk 1917
```

```
!
voice-port 1:17
 timeouts wait-release 3
 connection trunk 1918
!
voice-port 1:18
 timeouts wait-release 3
 connection trunk 1919
!
voice-port 1:19
 timeouts wait-release 3
 connection trunk 1920
!
voice-port 1:20
 vad
!
voice-port 1:21
 vad
!
voice-port 1:22
 vad
!
voice-port 1:23
 vad
!
dial-peer cor custom
!
dial-peer voice 19001 voip
 destination-pattern 5430001
 session target ipv4:172.16.0.0
 dtmf-relay cisco-rtp
 codec g711alaw
 fax rate 14400
!
dial-peer voice 19002 voip
 destination-pattern 5430002
 session target ipv4:172.16.0.0
 dtmf-relay cisco-rtp
 codec g726r16
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 19003 voip
 destination-pattern 5430003
 session target ipv4:172.16.0.0
 dtmf-relay cisco-rtp
 codec g711ulaw
 fax rate 14400
!
dial-peer voice 19004 voip
 destination-pattern 5430004
 session target ipv4:172.16.0.0
 dtmf-relay cisco-rtp
 codec g728
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 19005 voip
 destination-pattern 5430005
 session target ipv4:172.16.0.0
 dtmf-relay h245-alphanumeric
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
```

```
dial-peer voice 19006 voip
 destination-pattern 5430006
 session target ipv4:172.16.0.0
 dtmf-relay h245-alphanumeric
 codec g723ar53
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 19007 voip
 destination-pattern 5430007
 session target ipv4:172.16.0.0
 dtmf-relay h245-signal
 codec g729br8
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 19008 voip
 destination-pattern 5430008
 session target ipv4:172.16.0.0
 dtmf-relay h245-signal
 codec g726r32
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 19009 vofr
 destination-pattern 5430009
 session target Serial0 100
 dtmf-relay
 codec g726r16
 fax rate 14400
!
dial-peer voice 19010 vofr
 destination-pattern 5430010
 session target Serial0 100
 dtmf-relay
 codec g723ar63
 fax rate 14400
!
dial-peer voice 19011 vofr
 destination-pattern 5430011
 session target Serial0 100
 dtmf-relay
 codec g723r53
 fax rate 14400
!
dial-peer voice 19012 vofr
 destination-pattern 5430012
 session target Serial0 100
 dtmf-relay
 codec g723r63
 fax rate 14400
!
dial-peer voice 19101 vofr
 destination-pattern 5430013
 session target Serial0 100
 dtmf-relay
 codec g726r16
 fax rate 14400
!
dial-peer voice 19102 vofr
 destination-pattern 5430014
 session target Serial0 100
 dtmf-relay
 codec g726r24
```

```
 fax rate 14400
!
dial-peer voice 19103 vofr
 destination-pattern 5430015
 session target Serial0 100
 dtmf-relay
 codec g726r32
 fax rate 14400
!
dial-peer voice 19104 vofr
 destination-pattern 5430016
 session target Serial0 100
 dtmf-relay
 codec g728
 fax rate 14400
!
dial-peer voice 19105 voatm
 destination-pattern 1917
 session protocol aal2-trunk
 session target ATM0 pvc 1/100 20
 codec aal2-profile ITUT 1 g711ulaw
 dtmf-relay
!
dial-peer voice 19106 voatm
 destination-pattern 1918
 session protocol aal2-trunk
 session target ATM0 pvc 1/100 21
 dtmf-relay
 codec aal2-profile custom 110 g726r32
!
dial-peer voice 19107 voatm
 destination-pattern 1919
 session protocol aal2-trunk
 session target ATM0 pvc 1/100 22
 codec aal2-profile ITUT 1 g711ulaw
 dtmf-relay
!
dial-peer voice 19108 voatm
 destination-pattern 1920
 session protocol aal2-trunk
 session target ATM0 pvc 1/100 23
 dtmf-relay
 codec aal2-profile custom 110 g729br8
!
dial-peer voice 19109 voip
 destination-pattern 5430021
 session target ipv4:172.16.0.0
 codec g723r53
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 19110 voip
 destination-pattern 5430022
 session target ipv4:172.16.0.0
 codec g723r63
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 19111 voip
 destination-pattern 5430023
 session target ipv4:172.16.0.0
 codec g711alaw
 fax rate 14400
 fax protocol cisco
```

```
!
dial-peer voice 19112 voip
 destination-pattern 5430024
 session target ipv4:172.16.0.0
 codec g711ulaw
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 18001 pots
 destination-pattern 5420001
 port 1:0
!
dial-peer voice 18002 pots
 destination-pattern 5420002
 port 1:1
!
dial-peer voice 18003 pots
 destination-pattern 5420003
 port 1:2
!
dial-peer voice 18004 pots
 destination-pattern 5420004
 port 1:3
!
dial-peer voice 18005 pots
 destination-pattern 5420005
 port 1:4
!
dial-peer voice 18006 pots
 destination-pattern 5420006
 port 1:5
!
dial-peer voice 18007 pots
 destination-pattern 5420007
 port 1:6
!
dial-peer voice 18008 pots
 destination-pattern 5420008
 port 1:7
!
dial-peer voice 18009 pots
 destination-pattern 5420009
 port 1:8
!
dial-peer voice 18010 pots
 destination-pattern 5420010
 port 1:9
!
dial-peer voice 18011 pots
 destination-pattern 5420011
 port 1:10
!
dial-peer voice 18012 pots
 destination-pattern 5420012
 port 1:11
!
dial-peer voice 18101 pots
 destination-pattern 5420013
 port 1:12
!
dial-peer voice 18102 pots
 destination-pattern 5420014
 port 1:13
!
```

```
dial-peer voice 18103 pots
 destination-pattern 5420015
 port 1:14
!
dial-peer voice 18104 pots
 destination-pattern 5420016
 port 1:15
!
dial-peer voice 1817 pots
 destination-pattern 1817
 port 1:16
!
dial-peer voice 1818 pots
 destination-pattern 1818
 port 1:17
!
dial-peer voice 1819 pots
 destination-pattern 1819
 port 1:18
!
dial-peer voice 1820 pots
 destination-pattern 1820
 port 1:19
!
!
line con 0
 exec-timeout 0 0
line aux 0
line 2 3
line vty 0 4
 password lab
 login
!
end
```

The following is example **show running config** output from the terminating Cisco MC3810:

```
Router# show running config
!
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 3810eb-19-hc-atm
!
enable password lab
!
network-clock base-rate 56k
ip subnet-zero
ip domain-name cisco.com
ip host santa 172.16.0.0
ip name-server 172.16.0.1
ip name-server 172.16.0.2
!
isdn voice-call-failure 0
call rsvp-sync
!
no voice confirmation-tone
voice-card 0
 codec complexity high ecan-extended
!
controller T1 0
```

```
 mode atm
 framing esf
 clock source loop-timed
 linecode b8zs
!
controller T1 1
 mode cas
 framing esf
 linecode b8zs
 ds0-group 0 timeslots 1 type e&m-wink-start
 ds0-group 1 timeslots 2 type e&m-wink-start
 ds0-group 2 timeslots 3 type e&m-wink-start
 ds0-group 3 timeslots 4 type e&m-wink-start
 ds0-group 4 timeslots 5 type e&m-wink-start
 ds0-group 5 timeslots 6 type e&m-wink-start
 ds0-group 6 timeslots 7 type e&m-wink-start
 ds0-group 7 timeslots 8 type e&m-wink-start
 ds0-group 8 timeslots 9 type e&m-wink-start
 ds0-group 9 timeslots 10 type e&m-wink-start
 ds0-group 10 timeslots 11 type e&m-wink-start
 ds0-group 11 timeslots 12 type e&m-wink-start
 ds0-group 12 timeslots 13 type e&m-wink-start
 ds0-group 13 timeslots 14 type e&m-wink-start
 ds0-group 14 timeslots 15 type e&m-wink-start
 ds0-group 15 timeslots 16 type e&m-wink-start
 ds0-group 16 timeslots 17 type e&m-wink-start
 ds0-group 17 timeslots 18 type e&m-wink-start
 ds0-group 18 timeslots 19 type e&m-wink-start
 ds0-group 19 timeslots 20 type e&m-wink-start
 ds0-group 20 timeslots 21 type e&m-wink-start
 ds0-group 21 timeslots 22 type e&m-wink-start
 ds0-group 22 timeslots 23 type e&m-wink-start
 ds0-group 23 timeslots 24 type e&m-wink-start
!
interface Ethernet0
 ip address 172.29.251.19 255.255.255.0
 no ip route-cache
 no ip mroute-cache
!
interface Serial0
 bandwidth 2000000
 no ip address
 encapsulation frame-relay
 no ip route-cache
 no ip mroute-cache
 no keepalive
 frame-relay traffic-shaping
 frame-relay interface-dlci 100
  class fr200
  vofr cisco
 hold-queue 1024 out
!
interface Serial1
 no ip address
 no ip route-cache
 no ip mroute-cache
 shutdown
 no cdp enable
!
interface ATM0
 no ip address
 no atm ilmi-keepalive
 pvc 1/100
  vbr-rt 1536 1536 65535
```

```
  encapsulation aal2
 !
interface FR-ATM20
 no ip address
 no ip route-cache
 shutdown
!
ip classless
ip route 0.0.0.0 0.0.0.0 172.16.0.0
no ip http server
!
!
map-class frame-relay fr200
 frame-relay voice bandwidth 500000
 frame-relay fragment 80
 frame-relay traffic-rate 512000 1500000
 no frame-relay adaptive-shaping
 frame-relay cir 100000
 frame-relay mincir 100000
 frame-relay fair-queue
!
voice-port 1:0
!
voice-port 1:1
!
voice-port 1:2
!
voice-port 1:3
!
voice-port 1:4
!
voice-port 1:5
!
voice-port 1:6
!
voice-port 1:7
!
voice-port 1:8
!
voice-port 1:9
!
voice-port 1:10
!
voice-port 1:11
!
voice-port 1:12
!
voice-port 1:13
!
voice-port 1:14
!
voice-port 1:15
!
voice-port 1:16
 timeouts wait-release 3
 connection trunk 1817
!
voice-port 1:17
 timeouts wait-release 3
 connection trunk 1818
!
voice-port 1:18
 timeouts wait-release 3
 connection trunk 1819
```

```
!
voice-port 1:19
 timeouts wait-release 3
 connection trunk 1820
!
voice-port 1:20
!
voice-port 1:21
!
voice-port 1:22
!
voice-port 1:23
!
dial-peer cor custom
!
dial-peer voice 19001 pots
 destination-pattern 5430001
 port 1:0
 prefix ,,,5500001
!
dial-peer voice 19002 pots
 destination-pattern 5430002
 port 1:1
 prefix ,,,5500002
!
dial-peer voice 19003 pots
 destination-pattern 5430003
 port 1:2
 prefix ,,,5500003
!
dial-peer voice 19004 pots
 destination-pattern 5430004
 port 1:3
 prefix ,,,5500004
!
dial-peer voice 19005 pots
 destination-pattern 5430005
 port 1:4
 prefix ,,,5500005
!
dial-peer voice 19006 pots
 destination-pattern 5430006
 port 1:5
 prefix ,,,5500006
!
dial-peer voice 19007 pots
 destination-pattern 5430007
 port 1:6
 prefix ,,,5500007
!
dial-peer voice 19008 pots
 destination-pattern 5430008
 port 1:7
 prefix ,,,5500008
!
dial-peer voice 19009 pots
 destination-pattern 5430009
 port 1:8
 prefix ,,,5500009
!
dial-peer voice 19010 pots
 destination-pattern 5430010
 port 1:9
 prefix ,,,5500010
```

```
!
dial-peer voice 19011 pots
 destination-pattern 5430011
 port 1:10
 prefix ,,,5500011
!
dial-peer voice 19012 pots
 destination-pattern 5430012
 port 1:11
 prefix ,,,5500012
!
dial-peer voice 19101 pots
 destination-pattern 5430013
 port 1:12
 prefix ,,,5500013
!
dial-peer voice 19102 pots
 destination-pattern 5430014
 port 1:13
 prefix ,,,5500014
!
dial-peer voice 19103 pots
 destination-pattern 5430015
 port 1:14
 prefix ,,,5500015
!
dial-peer voice 19104 pots
 destination-pattern 5430016
 port 1:15
 prefix ,,,5500016
!
dial-peer voice 19105 pots
 destination-pattern 5430017
 port 1:16
 prefix ,,,5500017
!
dial-peer voice 19106 pots
 destination-pattern 5430018
 port 1:17
 prefix ,,,5500018
!
dial-peer voice 19107 pots
 destination-pattern 5430019
 port 1:18
 prefix ,,,5500019
!
dial-peer voice 19108 pots
 destination-pattern 5430020
 port 1:19
 prefix ,,,5500020
!
dial-peer voice 19109 pots
 destination-pattern 5430021
 port 1:20
 prefix ,,,5500021
!
dial-peer voice 19110 pots
 destination-pattern 5430022
 port 1:21
 prefix ,,,5500022
!
dial-peer voice 19111 pots
 destination-pattern 5430023
 port 1:22
```

```
 prefix ,,,5500023
!
dial-peer voice 19112 pots
 destination-pattern 5430024
 port 1:23
 prefix ,,,5500024
!
dial-peer voice 8888 pots
 destination-pattern 8888
!
dial-peer voice 18009 vofr
 incoming called-number 5430009
 destination-pattern 5420009
 session target Serial0 100
 dtmf-relay
 codec g726r16
 fax rate 14400
!
dial-peer voice 18010 vofr
 incoming called-number 5430010
 destination-pattern 5420010
 session target Serial0 100
 dtmf-relay
 codec g723ar63
 fax rate 14400
!
dial-peer voice 18011 vofr
 incoming called-number 5430011
 destination-pattern 5420011
 session target Serial0 100
 dtmf-relay
 codec g723r53
 fax rate 14400
!
dial-peer voice 18012 vofr
 incoming called-number 5430012
 destination-pattern 5420012
 session target Serial0 100
 dtmf-relay
 codec g723r63
 fax rate 14400
!
dial-peer voice 18001 voip
 incoming called-number 5430001
 destination-pattern 5420001
 session target ipv4:172.16.0.0
 dtmf-relay cisco-rtp
 codec g711alaw
 fax rate 14400
!
dial-peer voice 18002 voip
 incoming called-number 5430002
 destination-pattern 5420002
 session target ipv4:172.16.0.0
 dtmf-relay cisco-rtp
 codec g726r16
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 18003 voip
 incoming called-number 5430003
 destination-pattern 5420003
 session target ipv4:172.16.0.0
 dtmf-relay cisco-rtp
```

```
 codec g711ulaw
 fax rate 14400
!
dial-peer voice 18004 voip
 incoming called-number 5430004
 destination-pattern 5420004
 session target ipv4:172.16.0.0
 dtmf-relay cisco-rtp
 codec g728
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 18005 voip
 incoming called-number 5430005
 destination-pattern 5420005
 session target ipv4:172.16.0.0
 dtmf-relay h245-alphanumeric
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 18006 voip
 incoming called-number 5430006
 destination-pattern 5420006
 session target ipv4:172.16.0.0
 dtmf-relay h245-alphanumeric
 codec g723ar53
 fax rate 14400
 fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 18007 voip
 incoming called-number 5430007
 destination-pattern 5420007
 session target ipv4:172.16.0.0
 dtmf-relay h245-signal
 codec g729br8
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 18008 voip
 incoming called-number 5430008
 destination-pattern 5420008
 session target ipv4:172.16.0.0
 dtmf-relay h245-signal
 codec g726r32
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 18101 vofr
 incoming called-number 5430013
 destination-pattern 5420013
 session target Serial0 100
 dtmf-relay
 codec g726r16
 fax rate 14400
!
dial-peer voice 18102 vofr
 incoming called-number 5430014
 destination-pattern 5420014
 session target Serial0 100
 dtmf-relay
 codec g726r24
 fax rate 14400
!
dial-peer voice 18103 vofr
```

```
 incoming called-number 5430015
 destination-pattern 5420015
 session target Serial0 100
 dtmf-relay
 codec g726r32
 fax rate 14400
!
dial-peer voice 18104 vofr
 incoming called-number 5430016
 destination-pattern 5420016
 session target Serial0 100
 dtmf-relay
 codec g728
 fax rate 14400
!
dial-peer voice 18109 voip
 incoming called-number 5430021
 destination-pattern 5420021
 session target ipv4:172.29.251.18
 codec g723r53
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 18110 voip
 incoming called-number 5430022
 destination-pattern 5420022
 session target ipv4:172.29.251.18
 codec g723r63
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 18111 voip
 incoming called-number 5430023
 destination-pattern 5420023
 session target ipv4:172.29.251.18
 codec g711alaw
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 18112 voip
 incoming called-number 5430024
 destination-pattern 5420024
 session target ipv4:172.16.0.0
 codec g711ulaw
 fax rate 14400
 fax protocol cisco
!
dial-peer voice 18105 voatm
 incoming called-number 1917
 destination-pattern 1817
 session protocol aal2-trunk
  dtmf-relay
 session target ATM0 pvc 1/100 20
 codec aal2-profile ITUT 1 g711ulaw
!
dial-peer voice 18106 voatm
 incoming called-number 1918
 destination-pattern 1818
 session protocol aal2-trunk
 session target ATM0 pvc 1/100 21
 dtmf-relay
 codec aal2-profile custom 110 g726r32
!
dial-peer voice 18107 voatm
```

```
 incoming called-number 1919
 destination-pattern 1819
 session protocol aal2-trunk
 session target ATM0 pvc 1/100 22
 codec aal2-profile ITUT 1 g711ulaw
 dtmf-relay
 !
dial-peer voice 18108 voatm
 incoming called-number 1920
 destination-pattern 1820
 session protocol aal2-trunk
 session target ATM0 pvc 1/100 23
 dtmf-relay
 codec aal2-profile custom 110 g729br8
!
line con 0
 exec-timeout 0 0
line aux 0
line 2 3
line vty 0 4
 password lab
 login
!
end
```

Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS 7750 Example

The following example enables the G.168 extended EC on a Cisco 1700 series or a Cisco ICS7750. The extended EC is enabled by default when the **medium** keyword is used.

```
Router(config)# codec complexity medium
```

Changing Codec Complexity on the Cisco 7200 Series Example

The following example changes codec complexity on a Cisco 7200 series:

```
Router# configure terminal
Router(config)# dspint dspfarm 2/0
Router(config-dspfarm)# codec medium ecan-extended
```

Adjusting the Echo Canceller Size Example

The following example adjusts the size of the extended EC to 64 ms on Cisco 3600 series routers:

```
voice-port 1/0:0
echo-cancel enable
echo-cancel coverage 64
```

Worst-Case Echo Return Loss Example

The following example checks worst-case echo return loss configuration:

```
Router# show running-config
```

```
show run | begin voice-port
```

```
voice-port 0:D
echo-canceller erl worst-case 3
playout-delay mode fixed
no comfort-noise
!
```

Checking the Active Calls Example

The following is sample output from the **show call active voice** command. Important fields are highlighted in bold. (See the **show call active** command in the "Command Reference" section on page 62 for descriptions of the field names and values in the output.)

```
Router# show call active voice

Total call-legs:2

SetupTime=7587246 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=7587506
CallDuration=00:00:11
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=2
TransmitPackets=101
TransmitBytes=1991
ReceivePackets=550
ReceiveBytes=11000
VOIP:
ConnectionId[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
IncomingConnectionId[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
RemoteIPAddress=172.29.248.111
RemoteUDPPort=17394
RoundTripDelay=4 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=FALSE

SessionProtocol=cisco
SessionTarget=
OnTimeRvPlayout=10300
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=69 ms
ReceiveDelay=69 ms
LostPackets=0
EarlyPackets=0
```

```
LatePackets=0
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
SignalingType=ext-signal
CallerName=
CallerIDBlocked=False
 GENERIC:
SetupTime=7587246 ms
Index=2
PeerAddress=133001
PeerSubAddress=
PeerId=133001
PeerIfIndex=8
LogicalIfIndex=7
ConnectTime=7587505
CallDuration=00:00:56
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=2801
TransmitBytes=56020
ReceivePackets=162
ReceiveBytes=3192
TELE:
ConnectionId=[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
IncomingConnectionId=[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
TxDuration=56030 ms
VoiceTxDuration=3210 ms
FaxTxDuration=0 ms
CoderTypeRate=g729r8
NoiseLevel=-44
ACOMLevel=-13
OutSignalLevel=-45
InSignalLevel=-45
InfoActivity=2
ERLLevel=7
EchoCancellerMaxReflector=64
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
```

Additional References

For additional information related to the Enhanced ITU-T G.168 Echo Cancellation feature, see the following sections:

- Related Documents, page 61
- RFCs, page 61
- Standards, page 61
- MIBs, page 62
- Technical Assistance, page 62

Related Documents

Related Topic	Document Title
Cisco 1700 series routers	Cisco 1700 series routers documentation index
Cisco 2600 series routers	Cisco 2600 series routers documentation index
Cisco 2600 series hardware	Cisco 2600 Series Multiservice Platforms
Cisco 3600 series routers	Cisco 3600 series routers documentation index
Cisco 3600 series hardware	Cisco 3600 Series Multiservice Platforms
Cisco 3700 series routers	Cisco 3700 series routers documentation index
Cisco 7200 series routers	*Cisco 7200 Series*
Cisco 7500 series routers	*Cisco 7500 Series Routers*
How to configure your Cisco router or access server to support voice, video, and fax applications	*Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2T
How to use Cisco IOS commands to support voice, video, and fax applications	*Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2T
Cisco VG200	Cisco Voice Gateway 200 (VG200) release notes index
How to configure voice ports	*Configuring Voice Ports*, Cisco IOS Release 12.2
Cisco IOS software configuration	Configuration guides and command references, Cisco IOS Release 12.2T
Cisco IOS Release 12.2	Release notes index, Cisco IOS Release 12.2T
Echo analysis.	*Echo Analysis for Voice over IP*
Technical information organized by hardware products	*Hardware Support*
Cisco MC3810	Multiservice Access Concentrators documentation index
Configuration of VoIP on Cisco 2600 and Cisco 3600 series routers	Voice over IP for the Cisco 2600 and Cisco 3600 series routers documentation index

RFCs

RFCs	Title
No new or modified RFCs are supported by this feature, and support for existing RFCs has not been modified by this feature.	—

Standards

Standards[1]	Title
• ITU-T G.164	*Echo Suppressors*
• ITU-T G.165	*Echo Cancellers*
• ITU-T G.168 (04/2000)	*Digital Network Echo Cancellers*

1. Not all supported standards are listed.

MIBs

MIBs[1]	MIBs Link
• CISCO-VOICE-IF-MIB • CISCO-VOICE-DIAL-CONTROL-MIB	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

1. Not all supported MIBs are listed.

Technical Assistance

Description	Link
Technical Assistance Center (TAC) home page, containing 30,000 pages of searchable technical content, including links to products, technologies, solutions, technical tips, tools, and lots more. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/public/support/tac/home.shtml

Command Reference

This section documents new and modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.2T command reference publications.

Obsolete and Replaced Commands

Table 2 lists those commands that have been replaced since Cisco IOS Release 12.2(13)T.

Table 2 Replaced Extended Echo Canceller Commands

Command in Cisco IOS Release 12.2(13)ZH	Replacement Command in Cisco IOS Release 12.2(13)ZH
voice echo-canceller extended	**Obsolete for the Cisco 1700 series and the Cisco ICS 7750.**

New Commands

- **echo-cancel erl worst-case**
- **echo suppressor**
- **test call id**

Modified Commands

- **codec complexity**
- **comfort-noise**
- **destination-pattern**

- dial-peer voice
- ds0-group
- dspint dspfarm
- echo cancel coverage
- echo cancel enable
- non-linear
- port (dial peer)
- prefix
- show call active
- show voice call
- voice-card
- voice-port

codec complexity

To match the digital signal processor (DSP) complexity packaging to the codecs to be supported, use the **codec complexity** command in voice-card configuration mode. To reset to the default value, use the **no** form of this command.

codec complexity {**high** | **medium**} [**ecan-extended**]

no codec complexity

Syntax Description

high	High-complexity packaging. The **high** keyword selects a higher codec complexity if that is required to support a particular codec or combination of codecs. Each DSP supports two voice channels encoded in any of the following formats: • **g711alaw**—G.711 a-law 64,000 bps • **g711ulaw**—G.711 u-law 64,000 bps • **g723ar53—**G.723.1 Annex A 5300 bps • **g723ar63—**G.723.1 Annex A 6300 bps • **g723r53**—G.723.1 5300 bps • **g723r63**—G.723.1 6300 bps • **g723r16**—G.726 16,000 bps • **g726r24**—G726 24,000 bps • **g726r32**—G.726 32,000 bps • **g728—**G.728 16,000 bps • **g729r8**—G.729 8000 bps (default) • **g729br8**—G.729 Annex B 8000 bps • **fax relay—**2400 bps, 4800 bps, 7200 bps, 9600 bps, 12 kbps, and 14.4 kbps **Note** Codecs G.723.1 and G.728 are not supported on Cisco 1750 and Cisco 1751 for Cisco Hoot and Holler over IP applications.

medium	Medium-complexity packaging. The **medium** keyword selects a lower codec complexity to support the greatest number of voice channels, provided that the lower complexity is compatible with the particular codecs in use. Each DSP supports four voice channels encoded in any of the following formats: • **g711alaw**—G.711 a-law 64,000 bps • **g711ulaw**—G.711 u-law 64,000 bps • **g726r16**—G.726 16,000 bps • **g726r24**—G.726 24,000 bps • **g726r32**—G.726 32,000 bps • **g729r8**—G.729 Annex A 8000 bps • **G729br8**—G.729 Annex B with Annex A 8000 bps • **fax relay**—(default) 2400 bps, 4800 bps, 7200 bps, 9600 bps, 12 kbps, and 14.4 kbps
ecan-extended	(Optional) Selects the extended echo canceller. Use this keyword when either the **codec complexity high** or the **codec complexity medium** options are chosen. The default option is to use the Cisco proprietary G.165-compliant echo canceller (EC).

Defaults

Medium-complexity codecs

Command Modes

Voice-card configuration

Command History

Release	Modification
12.0(5)XK	This 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 for use with the high-performance compression module (HCM).
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(8)T	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(13)T	The **ecan-extended** keyword was added.
12.2(13)ZH	The extended G.168 EC is the default on the Cisco 1700 series and the Cisco 7750 when the **medium** keyword is used. The **ecan-extended** keyword is not used on these platforms.

Usage Guidelines

Codec complexity refers to the amount of processing required to perform voice compression. Codec complexity affects call density—the number of calls that the DSPs can handle. With higher codec complexity, fewer calls can be handled. Select a higher codec complexity if that is required to support a particular codec or combination of codecs. Select a lower codec complexity to support the greatest number of voice channels, provided that the lower complexity is compatible with the particular codecs in use.

Before you change codec complexity, you must place all of the DSP voice channels in the idle state.

Note In the Cisco MC3810, this command is valid only with installed HCMs, and you must specify voice card 0. If two HCMs are installed, the **codec complexity** command configures both HCMs at once.

You can construct two separate configurations, one for the Cisco default EC and one for the extended EC, which you can load manually by creating new configurations for each type of EC and reloading the router.

- Use the **codec complexity high** command for the Cisco default EC.
- Use the **codec complexity high ecan-extended** command for the extended EC.
- Use the **codec complexity medium ecan-extended** command for the extended EC.
- Use the **codec complexity medium** command to enable the extended EC on the Cisco 1700 series and the Cisco ICS 7750.

Examples

The following example sets the codec complexity to high on a Cisco MC3810 that contains one or two HCMs:

```
Router(config)# voice-card 0
Router(config-voicecard)# codec complexity high
```

The following example sets the codec complexity to high on voice card 1 in a Cisco 2600 or Cisco 3600 series router:

```
Router(config)# voice-card 1
Router(config-voicecard)# codec complexity high
```

The following example changes the codec complexity:

```
Router(voice-card)# codec complexity high ecan-extended
```

Related Commands

Command	Description
ds0-group	Defines T1/E1 channels for compressed voice calls and the CAS method by which the router connects to the PBX or PSTN.
show voice dsp	Shows the current status of all DSP voice channels.

comfort-noise

To generate background noise to fill silent gaps during calls if voice activity detection (VAD) is enabled, use the **comfort-noise** command in voice-port configuration mode. To provide silence, use the **no** form of this command.

comfort-noise

no comfort-noise

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T, and was implemented on the Cisco 2600 series, Cisco 7200 series, and Cisco 7500 series using the extended echo canceller.

Usage Guidelines

If this command is not enabled and VAD is enabled at the remote end of the connection, the user hears dead silence when the remote party is not speaking.

The configuration of this command affects only the silence generated at the local interface; it does not affect the use of VAD on either end of the connection or the silence generated at the remote end of the connection.

Note On the Cisco MC3810, this command cannot be disabled.

Examples

The following example enables background noise on the Cisco 3600 series:

```
Router(voice-port)# comfort-noise
```

```
 voice-port
  comfort-noise
```

Related Commands

Command	Description
vad (dial-peer configuration)	Enables VAD for calls using a particular dial peer.
vad (voice-port configuration)	Enables VAD for calls using a particular voice port on the Cisco MC3810.

destination-pattern

To specify either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer, use the **destination-pattern** command in dial-peer configuration mode. To disable the prefix or telephone number, use the **no** form of this command.

destination-pattern [+] *string* [**T**]

no destination-pattern [+] *string* [**T**]

Syntax Description

+	(Optional) Character that indicates an E.164 standard number.
string	Series of digits that specify the E.164 or private-dialing-plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters: • The asterisk (*) and pound sign (#) that appear on standard touchtone dial pads. On the Cisco 3600 series routers only, these characters cannot be used as leading characters in a string (for example, *650). • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit (this character is used as a wildcard). On the Cisco 3600 series routers, the period cannot be used as a leading character in a string (for example, .650). • Percent sign (%), which indicates that the previous digit or pattern occurred zero or multiple times, similar to the wildcard usage in the regular expression. • Plus sign (+), which matches a sequence of one or more matches of the character or pattern. **Note** The plus sign used as part of the digit string is different from the plus sign that can be used in front of the digit string to indicate that the string is an E.164 standard number. • Circumflex (^), which indicates a match to the beginning of the string. • Dollar sign ($), which indicates a match the null string at the end of the input string. • Backslash symbol (\), followed by a single character matching that character or used with a single character with no other significance (matching that character). • Question mark (?), which indicates that the previous digit occurred zero or one time. • Brackets ([ ]), which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range. This is similar to a regular expression rule. • Parentheses "( )", which indicate a pattern and are the same as the regular expression rule.
T	(Optional) Control character that indicates that the **destination-pattern** value is a variable-length dial string.

Defaults

Enabled with a null string.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(1)MA	This command was implemented on the Cisco MC3810.
12.0(4)XJ	This command was modified for store-and-forward fax.
12.1(1)	The command as modified for store-and-forward fax was integrated into Cisco IOS Release 12.1(1).
12.0(7)XR	Support for the plus sign, percent sign, question mark, brackets, and parentheses symbols in the dial string was added to the Cisco AS5300.
12.0(7)XK	Support for the plus sign, percent sign, question mark, brackets, and parentheses in the dial string was added to the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.1(1)T	Modifications made in Cisco IOS Release 12.0(7)XR for the Cisco AS5300 were integrated into Cisco IOS Release 12.1(1)T and were implemented on the Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco 7200, and Cisco 7500 series.
12.1(2)T	Modifications made in Cisco IOS Release 12.0(7)XK for the Cisco MC3810 were first supported on the T train.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T and implemented on the Cisco 2600XM, Cisco ICS7750, and Cisco VG200.

Usage Guidelines

This pattern created by this command is used to match dialed digits to a dial peer. The dial peer is then used to complete the call. When a router receives voice data, it compares the called number (the full E.164 telephone number) in the packet header with the number configured as the destination pattern for the voice-telephony peer. The router then strips out the left-justified numbers corresponding to the destination pattern. If you have configured a prefix, the prefix is appended to the front of the remaining numbers, creating a dial string, which the router then dials. If all numbers in the destination pattern are stripped out, the user receives a dial tone.

There are certain areas in the world (for example, certain European countries) where valid telephone numbers can vary in length. Use the optional control character **T** to indicate that a particular **destination-pattern** value is a variable-length dial string. In this case, the system does not match the dialed numbers until the interdigit timeout value has expired.

Note Cisco IOS software does not check the validity of the E.164 telephone number; it accepts any series of digits as a valid number.

Examples

The following example shows configuration of the E.164 telephone number 555-7922 for a dial peer:

```
dial-peer voice 10 pots
 destination-pattern +5557922
```

The following example shows configuration of a destination pattern in which the pattern "43" is repeated multiple times after the digits "555":

```
dial-peer voice 1 voip
 destination-pattern 555(43)+
```

The following example shows configuration of a destination pattern in which the preceding digit or pattern is repeated multiple times:

```
dial-peer voice 2 voip
 destination-pattern 555%
```

The following example shows configuration of a destination pattern in which the digit numbers range from between 5553409 to 5559499:

```
dial-peer voice 3 vofr
 destination-pattern 555[3-9]4[0=9]9
```

The following example shows configuration of a destination pattern in which the digit numbers range from between 5551439, 5553439, 5555439, 5557439, and 5559439:

```
dial-peer voice 4 voatm
 destination-pattern 555[13579]439
```

Related Commands

Command	Description
answer-address	Specifies the full E.164 telephone number to be used to identify the dial peer of an incoming call.
prefix	Specifies the prefix of the dialed digits for this dial peer.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.

dial-peer voice

To define a particular dial peer to specify the method of voice encapsulation and to enter dial-peer configuration mode, use the **dial-peer voice** command in global configuration mode. To disable a defined dial peer, use the **no** form of this command. Alternately, you can disable a dial peer using the **no shutdown** command in dial-peer configuration mode.

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

Cisco 1750 and Cisco 1751

dial-peer voice *tag* {**pots** | **vofr** | **voip**}

no dial-peer voice *tag* {**pots** | **vofr** | **voip**}

Cisco 2600 Series, Cisco 2600XM, CIsco 3600 Series, Cisco 3700 Series, Cisco IAD2420 Series, and Cisco VG200

dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

no dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

Cisco 7200 Series

dial-peer voice *tag* {**vofr**}

no dial-peer voice *tag* {**vofr**}

Cisco 7204 VXR and Cisco 7206 VXR

dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

no dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

Cisco AS5300

dial-peer voice *tag* {**mmoip** | **pots** | **vofr** | **voip**}

no dial-peer voice *tag* {**mmoip** | **pots** | **vofr** | **voip**}

Cisco MC3810

dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

no dial-peer voice *tag* {**pots** | **voatm** | **vofr** | **voip**}

Syntax Description

tag	Digits that define a particular dial peer. Valid entries are from 1 to 2147483647. There is no default.
mmoip	Multimedia mail peer using IP encapsulation on the IP backbone. **Note** On the Cisco AS5300, Multimedia Mail over IP (MMoIP) is available only if you have modem ISDN channel aggregation (MICA) technologies modems.
pots	Plain old telephone service (POTS) peer using Voice over IP encapsulation on the IP backbone.
voatm	(Cisco 2600 series, Cisco 3600 series, Cisco MC3810, Cisco 7204 VXR routers, and Cisco 7206 VXR only) Voice over ATM (VoATM) dial peer using real-time AAL5 voice encapsulation on the ATM backbone network.
vofr	Voice over Frame Relay (VoFR) dial peer using FRF.11 encapsulation on the Frame Relay backbone network.
voip	Voice over IP (VoIP) peer using voice encapsulation on the POTS network.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(1)MA	This command was implemented on the Cisco MC3810 with support for the **pots**, **voatm**, **vofr**, and **vohdlc** keywords.
12.0(3)T	This command was implemented on the Cisco AS5300 with support for the **pots** and **voip** keywords.
12.0(3)XG	The **vofr** keyword was added for the Cisco 2600 series and Cisco 3600 series.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T. The **vofr** keyword was added for the Cisco 7200 series.
12.0(4)XJ	The **mmoip** keyword was added for the Cisco AS5300. The **dial-peer voice** command was implemented for store-and-forward fax.
12.0(7)XK	The **voip** keyword was added for the Cisco MC3810. The **voatm** keyword was added for the Cisco 3600 series. Support for the **vohdlc** keyword on the Cisco MC3810 was removed in this release.
12.1(1)	The **mmoip** keyword addition in Cisco IOS Release 12.0(4)XJ was integrated into Cisco IOS Release 12.1(1). The **dial-peer voice** implementation for store-and-forward fax was also integrated into this mainline release.
12.1(2)T	The keyword changes in Cisco IOS Release 12.0(7)XK were integrated into Cisco IOS Release 12.1(2)T.

Release	Modification
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2. This command was implemented on the Cisco IAD2420 series.
12.2(13)T	This command was implemented on the Cisco 2600XM, Cisco ICS7750, and Cisco VG200.

Usage Guidelines

Use this command to switch to dial-peer configuration mode from global configuration mode. Use the **exit** command to exit dial-peer configuration mode and return to global configuration mode.

After you have created a dial peer, that dial peer remains active until you disable it. To disable a defined dial peer, use the **no** form of this command. Alternately, you can use the **no shutdown** command in dial-peer configuration mode.

In store-and-forward fax on the Cisco AS5300, the POTS dial peer defines the inbound-fax-line characteristics from the sending fax device to the receiving Cisco AS5300 and the outbound-line characteristics from the sending Cisco AS5300 to the receiving fax device. The Multimedia Mail over IP (MMoIP) dial peer defines the inbound-fax-line characteristics from the Cisco AS5300 to the receiving Simple Mail Transfer Protocol (SMTP) mail server. This command applies to both on-ramp and off-ramp store-and-forward fax functions.

Note On the Cisco AS5300, MMoIP is available only if you have Cisco MICA modems. The Cisco AS5300 does not support the extended EC feature.

Examples

The following example configures the extended echo canceller. In this instance, **pots** indicates that this is a plain old telephone service (POTS) peer using VoIP encapsulation on the IP backbone, and it uses the unique numeric identifier tag 133001.

```
Router(config)# dial-peer voice 133001 pots
```

The following example configures POTS peer identified dial peer 10 and MMoIP dial peer 20:

```
dial-peer voice 10 pots
dial-peer voice 20 mmoip
```

The following example deletes the MMoIP dial peer 20:

```
no dial-peer voice 20 mmoip
```

Related Commands

Command	Description
codec (dial peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
destination-pattern	Specifies the prefix, the full E.164 telephone number, or an ISDN directory number to be used for a dial peer.
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.
exit	Exits dial-peer configuration mode and returns to global configuration mode.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
sequence-numbers	Enables the generation of sequence numbers in each frame generated by the DSP for Voice over Frame Relay applications.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
shutdown	Disables a dial peer in dial-peer configuration mode.
voice-port	Enters voice-port configuration mode.

ds0-group

To specify the DS-0 time slots that make up a logical voice port on a T1 or E1 controller, to specify the signaling type by which the router communicates with the PBX or PSTN, and to define T1or E1 channels for compressed voice calls and the channel-associated signaling (CAS) method by which the router connects to the PBX or PSTN, use the **ds0-group** command in controller configuration mode. To remove the group and signaling setting, use the **no** form of this command.

Cisco 1750 and Cisco 1751 Routers —T1 and E1

ds0-group *ds0-group* **timeslots** *timeslot-list* **type** [**service** *service-type*{**data** | **fax** | **voice**} {**e&m-fgb** | **e&m-fgd** | **e&m-immediate-start** | **fgd-eana** | **fgd-os** | **fxs-ground-start** | **fxs-loop-start** | **none** | **r1-itu** | **r1-modified** | **r1-turkey** | **sas-ground-start** | **sas-loop-start**}]

no ds0-group *ds0-group*

Cisco 2600 Series, Cisco 3600 Series, and the Cisco MC3810—T1

ds0-group *ds0-group* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-fgd** | **e&m-immediate-start** | **e&m-wink-start** | **ext-sig** | **fgd-eana** | **fxo-ground-start** | **fxo-loop-start** | **fxs-ground-start** | **fxs-loop-start**}

no ds0-group *ds0-group*

Cisco 2600 Series, Cisco 3600 Series, and the Cisco MC3810—E1

ds0-group *ds0-group* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-melcas-delay** | **e&m-melcas-immed** | **e&m-melcas-wink** | **e&m-wink-start** | **ext-sig** | **fgd-eana** | **fxo-ground-start** | **fxo-loop-start** | **fxo-melcas** | **fxs-ground-start** | **fxs-loop-start** | **fxs-melcas** | **r2-analog** | **r2-digital** | **r2-pulse**}

no ds0-group *ds0-group*

Cisco 7200 Series and Cisco 7500 Series—T1 and E1 Voice Ports

ds0-group *ds0-group* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-fgd** | **e&m-immediate-start** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

no dso-group *ds0-group*

Cisco 7700 Series—T1 and E1 Voice Ports

ds0-group *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-immediate-start** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

no ds0-group *ds0-group*

Note Keywords for this command are configuration-specific. For example, if MGCP is configured, you see the **mgcp** keyword. If MGCP is not configured, you do not see the **mgcp** keyword.

In addition, keywords for this command are dependent upon the Cisco IOS release that you are using. Refer to Cisco Feature Navigator at the following URL for information regarding your release:

http://www.cisco.com/go/fn

Syntax Description

addr info	(Optional) Specifies the calling or called party.
ds0-group	A DS-0 group number. T1 range is from 0 to 23. E1 range is from 0 to 14 and 16 to 30;15 is reserved.
service *service-type*	(Optional) Type of service. • **data**—data service • **fax**— store-and-forward fax service • **voice**—voice service
timeslots *timeslot-list*	A single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas. T1 range is from 1 to 24. E1 range is from 1 through 31. The following are examples: • 2 • 1-15,17-24 • 1-23 • 2,4,6-12

tone *type*	(Optional) Tone type: either **dtmf** or **mf**.

type	Signaling method selection for the **type** keyword depends on the connection that you are making. The E&M interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The FXS interface allows connection of basic telephone equipment and PBX. The FXO interface allows connection of the central office (CO) to a standard PBX interface where permitted by local regulations; it is often used for off-premise extensions (OPXs). Types are as follows: • **e&m-delay-dial**—The originating endpoint sends an off-hook signal and then waits for an off-hook signal followed by an on-hook signal from the destination • **e&m-fgb**—E&M Type II Feature Group B • **e&m-fgd**—E&M Type II Feature Group D • **e&m-immediate-start**—E&M immediate start • **e&m-melcas-delay**—E&M MELCAS delay-start signaling support • **e&m-melcas-immed**—E&M MELCAS immediate-start signaling support • **e&m-melcas-wink**—E&M MELCAS wink-start signaling support • **e&m-wink-start**—The originating endpoint sends an off-hook signal and waits for a wink-start from the destination • **ext-sig**—An option available only when the **mode CCS** command is enabled on the Cisco MC3810 for FRF.11 transparent CCS support • **fgd-eana**—Feature Group D exchange access North American • **fgd-os**—Feature Group D operator services • **fxo-ground-start**—FXO ground-start signaling • **fxo-loop-start**—FXO loop-start signaling • **fxo-melcas**—FXO MELCAS signaling • **fxs-ground-start**—FXS ground-start signaling • **fxs-loop-start**—FXS loop-start signaling • **fxs-melcas**—FXS MELCAS signaling • **none**—Null signaling for external call control • **p7**—p7 switch type • **r1-itu**—Line signaling based on international signaling standards • **r1-modified**—An international signaling standard that is common to channelized T1/E1 networks • **r1-turkey**—A signaling standard used in Turkey • **r2-analog**—R2 analog line signaling • **r2-digital**—R2 digital line signaling • **r2-lsv181-digital**—A specific R2 digital line • **r2-pulse**—7-pulse line signaling, a transmitted pulse that indicates a change in the line state • **sas-ground-start**—Single attachment station (SAS) ground-start • **sas-loop-start**—SAS loop-start

Defaults

No DS-0 group is defined. Calls are allowed in both directions.

Command Modes

Controller configuration

Command History

Release	Modification
11.2	This command was introduced on the Cisco AS5300 as the **cas-group** command.
11.3(1)MA	The command was introduced on the Cisco MC3810 as the **voice-group** command.
12.0(1)T	The **cas-group** command was implemented on the Cisco 3600 series routers.
12.0(5)T	The command was renamed **ds0-group** on the Cisco AS5300, Cisco 2600 series. and Cisco 3600 series. Some keyword modifications were made.
12.0(5)XE	This command was implemented on the Cisco 7200 series.
12.0(7)XK	Support for this command was extended to the Cisco MC3810. When this command became available on the Cisco MC3810, the **voice-group** command was removed and no longer supported. The **ext-sig** keyword replaced the **ext-sig-master** and **ext-sig-slave** keywords.
12.0(7)XR	The **mgcp** service type was added.
12.1(1)T	This command was implemented on the Cisco 7200 series.
12.1(2)XH	The **e&m-fgd** and **fgd-eana** keywords were added for Feature Group D signaling.
12.1(3)T	This command was implemented on the Cisco 7500 series. The **fgd-os** signaling type and the **voice** service type were added.
12.2(2)XA	This command was implemented on the Cisco AS5300.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T and implemented on the Cisco 7200 series.
12.2(4)T	Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751. Support for other Cisco platforms is not included in this release.
12.2(2)XN	Support for the **mgcp** keyword was added to Cisco CallManager Version 3.1 for the Cisco 2600 series, Cisco 3600 series, and Cisco VG200.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.

Release	Modification
12.2(11)T	This command was supported with Cisco IOS Release 12.2(11)T and Cisco CallManager Version 3.2. This command is supported on the Cisco IAD2420 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T. The Cisco 1750 and Cisco 1751 do not support T1 and E1 voice and data cards in Cisco IOS Release 12.2(13)T. The Cisco 17xx platforms support only HC DSP firmware images in this release.

Usage Guidelines

This command automatically creates a logical voice port that is numbered as follows:

- Cisco 2600, Cisco 3600, and Cisco 7200 series: *slot*/*port***:***ds0-group*
- Cisco MC3810: *slot***:***ds0-group*

  On the Cisco MC3810, the *slot* number is the controller number.
- Cisco AS5300 with a T1 controller: *slot*/*port*

Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

Note This command does not support the extended echo canceller (EC) feature on the Cisco AS5x00 series.

Examples

The following example configures ranges of T1 controller time slots for FXS ground-start and FXO loop-start signaling on a Cisco 2600 series, Cisco 3600 series, or Cisco 3700 series:

```
T1 1/0
 framing esf
 linecode b8zs
 ds0-group 1 timeslots 1-10 type fxs-ground-start
 ds0-group 2 timeslots 11-24 type fxo-loop-start
```

The following example configures DS-0 groups 1 and 2 on controller T1 1 on the Cisco MC3810 to support transparent common channel signaling (CCS):

```
controller T1 1
 mode ccs cross-connect
 ds0-group 1 timeslots 1-10 type ext-sig
 ds0-group 2 timeslots 11-24 type ext-sig
```

The following example configures ranges of T1 controller time slots for FXS ground-start signaling:

```
controller T1 1/0
 ds0-group 1 timeslots 1-4 type fxs-ground-start
```

The following example set the T1 channels for SS7 service on any trunking gateway in **mgcp** mode:

```
 ds0-group 0 timeslots 1-24 type none service mgcp
```

The following example sets the T1 channels for SS7 service on any trunking gateway in **sgcp** mode:

```
 ds0-group 0 timeslots 1-24 type none service sgcp
```

The following example sets the T1 channels for FGD-OS service on an Cisco AS5300 in **sgcp** mode:

```
Router(config-controller)# ds0-group 0 timeslots 1-24 type fgd-os mf dnis-ani service
```

Related Commands

Command	Description
cas-group	Configures channelized T1 time slots with robbed-bit signaling.
codec (dial peer)	Specifies the voice coder rate of speech for a dial peer.
codec complexity	Specifies call density and codec complexity based on the codec standard that you are using.

dspint dspfarm

To enable the digital signal processor (DSP) interface, use the **dspint dspfarm** command in global configuration mode.

dspint dspfarm *slot*/*port*

Syntax Description

slot	Slot number of the interface.
port	Port number of the interface.

Defaults

Enabled by default

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XE	This command was introduced on the Cisco 7200 series routers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.
12.2(13)T	This command was implemented on the Cisco 7200 series.

Usage Guidelines

DSP mapping occurs when DSP resources on one advanced interface module (AIM) or network module are available for processing of voice time-division multiplexing (TDM) streams on a different network module or on a voice/WAN interface card (VWIC). This command is used on Cisco 3660 routers with multiservice interchange (MIX) modules installed or on Cisco 2600 series routers with AIMs installed.

The assignment of DSP pool resources to particular TDM streams is based on the order in which the streams are configured using the **ds0-group** command for T1/E1 channel-associated signaling (CAS) or using the **pri-group** command for ISDN PRI.

The assignment of DSP pool resources does not occur dynamically during call signaling.

To disable the DSPfarm interface, enter the **no shutdown** command.

Examples

The following example creates a DSPfarm interface with a slot number of 1 and a port number of 0.

```
dspint dspfarm 1/0
```

Related Commands

Command	Description
no shutdown	Changes the administrative state of a port from out of service to in service.
show interfaces dspfarm dsp	Displays information about the DSP interface.

echo cancel coverage

To adjust the size of the echo canceller (EC) and to select the extended EC when the Cisco default EC is present, use the **echo cancel coverage** command in voice-port configuration mode. To reset to the default value, use the **no** form of this command.

echo cancel coverage {**8** | **16** | **24** | **32** | **48** | **64**}

no echo cancel coverage

Syntax Description

8	EC size of 8 ms
16	EC size of 16 ms
24	EC size of 24 ms
32	EC size of 32 ms
48	EC size of 48 ms
64	EC size of 64 ms. This is the default

Defaults

64

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(1)MA	This command was implemented on the Cisco MC3810.
12.0(5)XK	This command was modified to add the 8-ms option.
12.0(5)XE	The command was implemented on the Cisco 7200 series.
12.1(1)T	The Cisco IOS Release 12.0(5)XK and Release 12.0(5)XE changes were integrated into Cisco IOS Release 12.1(1)T.
12.2(13)T	This command was modified to provide a new set of size options when the extended EC is configured. This command is supported on all T1 digital signal processor (DSP) platforms.

Usage Guidelines

Use this command to adjust the coverage size of the EC. This command enables cancellation of voice that is sent out the interface and received on the same interface within the configured amount of time. If the local loop (the distance from the interface to the connected equipment that is producing the echo) is greater than this amount of time, you should increase the configured value of this command.

If you configure a large value for this command, the EC takes longer to converge and you might hear a slight echo when the connection is initially set up. If you configure a small value, you might hear some echo for the duration of the call because the EC is not canceling the longer delay echoes.

There is no echo or echo cancellation on the network (for example, the non-POTS) side of the connection.

Note This command is valid only when the echo canceller feature is enabled. The Cisco proprietary G.165 EC is enabled by default. For more information, refer to the **echo cancel enable** command reference page.

Examples

The following example enables extended echo cancellation and adjusts the size of the echo canceller to 16 ms on the Cisco 3600 series:

```
voice-port 1/0/0
 echo cancel enable
 echo cancel coverage 16
```

The following example enables extended echo cancellation and adjusts the size of the echo canceller to 16 ms on the Cisco MC3810:

```
voice-port 1/1
 echo cancel enable
 echo cancel coverage 16
```

Related Commands

Command	Description
echo cancel enable	Enables the cancellation of voice that is sent out the interface and received on the same interface.

echo cancel enable

To enable cancellation—that is, cancellation of voice that is sent out and received on the same interface—use the **echo cancel enable** command in voice-port configuration mode. To disable echo cancellation, use the **no** form of this command.

echo cancel enable

no echo cancel enable

Syntax Description

This command has no arguments or keywords.

Defaults

The Cisco proprietary G.165 echo canceller (EC) is enabled with the echo suppressor turned off.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series and Cisco MC3810.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T and implemented on the Cisco 7200 series and Cisco 7500 series. This command is supported on all TI Digital Signal Processor (DSP) platforms.

Usage Guidelines

This command enables cancellation of voice that is sent out the interface and received back on the same interface; sound that is received back in this manner is perceived by the listener as an echo. Disabling echo cancellation might cause the remote side of a connection to hear an echo. Because echo cancellation is an invasive process that can minimally degrade voice quality, this command should be disabled if it is not needed.

Typically a hybrid circuit can provide greater than 6 decibels (dB) of echo return loss (ERL), so the extended EC is configured to handle 6 dB worst case by default. However, if a measurement shows that a circuit can provide only 6 dB ERL or less, you can configure the extended EC to use this lower rate.

The Cisco G.165 EC is enabled by default with the echo suppressor turned off. The echo suppressor can be turned on only with the default Cisco G.165 EC. The **echo suppressor** command used with the Cisco default EC is still visible when the extended EC is selected, but it does not do anything.

This command does not affect echo heard by the user on the analog side of the connection.

There is no echo path for a four-wire recEive and transMit (also called ear and mouth, abbreviated E&M) interface. Disable the echo canceller for that interface type.

Note This command is valid only when used with the **echo cancel coverage** command.

Examples

The following example enables extended echo cancellation and adjusts the size of the echo canceller to 16 ms on the Cisco 3600 series:

```
voice-port 1/0:0
 echo cancel enable
 echo cancel coverage 16
```

The following example enables extended echo cancellation and adjusts the size of the echo canceller to 16 ms on the Cisco MC3810:

```
voice-port 1/1
 echo cancel enable
 echo cancel coverage 16
```

The following example enables extended echo cancellation on the Cisco 1700 series or Cisco ICS7750:

```
Router(voice-card)# codec complexity medium
```

Related Commands

Command	Description
echo cancel coverage	Adjusts the size of the echo canceller.
non-linear	Enables nonlinear processing in the echo canceller.

echo-cancel erl worst-case

To predict the worst-case echo return loss (ERL) that the echo canceller might encounter, use the **echo cancel erl worst-case** command in voice-port configuration mode. To disable the command, use the **no** form.

echo cancel erl worst-case {**6** | **3** | **0**}

no echo cancel erl worst-case {**6** | **3** | **0**}

Syntax Description

6 | 3 | 0	Loss, in dB. Default is 6.

Defaults

6

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

This command is used only when the extended EC is present and is not supported with the Cisco proprietary G.165 EC. This command predicts the worst-case echo return loss that the EC might encounter.

To check the configuration, use the **show running-config** command in privileged EXEC mode.

Examples

The following example shows worst-case ERL selections:

```
Router(config-voiceport)# echo cancel erl worst-case ?

  0  Worst case echo canceller operation is 0 dB ERL
  3  Worst case echo canceller operation is 3 dB ERL
  6  Worst case echo canceller operation is 6 dB ERL
```

The following example sets the worst-case ERL to 3:

```
Router(config-voiceport)# echo cancel erl worst-case 3

22:51:13:%SYS-5-CONFIG_I:Configured from console by console run | begin
voice-port
```

Related Commands

Command	Description
echo cancel enable	Enables the cancellation of echo—that is, voice that is sent out and received on the same interface.

echo suppressor

To enable echo suppression to reduce initial echo before the echo canceller converges, use the **echo suppressor** command in voice-port configuration mode. To disable echo suppression, use the **no** form of this command.

echo suppressor *seconds*

no echo suppressor

Syntax Description

seconds	Suppression coverage, in seconds. Range is from 1 to 10. Default is 7.

Defaults

Disabled

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

Use this command only when the echo canceller is enabled. In case of double-talk in the first number of seconds, the code automatically disables the suppressor.

Examples

The following example configures echo suppression for a suppression coverage of 9 seconds on a Cisco 3620:

```
Router(config)# voice-port 1/1:0
Router(config-voiceport)# echo suppressor 9
```

Related Commands

Command	Description
echo cancel enable	Enables cancellation of echo—that is, voice that is sent out and is received on the same interface.

non-linear

To enable nonlinear processing in the echo canceller, use the **non-linear** command in voice-port configuration mode. To disable nonlinear processing, use the **no** form of this command.

non-linear

no non-linear

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T and implemented on the following platforms that support the extended G.168 echo canceller (EC): Cisco 1700 series, Cisco 2600 series, Cisco 2600XM, Cisco 3600 series, Cisco 3700 series, Cisco 7200 series, Cisco 7500 series, Cisco ICS7750, Cisco MC3810, and Cisco VG200.

Usage Guidelines

The function enabled by this command is also generally known as residual echo suppression. This command is associated with echo-canceller operation. The **echo cancel enable** command must be enabled for this command to take effect. Use this command to shut off any signal if no near-end speech is detected.

The Cisco G.165 EC is enabled by default with the echo suppressor disabled. You can enable the echo suppressor only when the default Cisco G.165 EC is used. The default echo suppressor command is still visible when the extended EC is selected, but it does not do anything.

Using this command normally improves performance, although some users might perceive truncation of consonants at the end of sentences when this command is enabled.

Examples

The following example shows that nonlinear call processing is enabled on a Cisco 3600 series:

```
voice-port 1/0/0
 non-linear
```

The following example shows that nonlinear call processing is enabled on a Cisco MC3810:

```
voice-port 1/1
 non-linear
```

Related Commands

Command	Description
echo cancel enable	Enables the cancellation of voice that is sent out and received on the same interface.

port (dial peer)

To associate a dial peer with a specific voice port, use the **port** command in dial-peer configuration mode. To cancel this association, use the **no** form of this command.

Cisco 1750 and Cisco 3700 Series

port *slot***/***port*

no port *slot***/***port*

Cisco 2600 and Cisco 3600 Series,

port {*slot-number***/***subunit***/***port* | *slot***/***port:ds0-group*}

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

Cisco 7200 Series

port {*slot/port:ds0-group* | *slot***/***subunit***/***port*}

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

Cisco MC3810

port *slot***/***port*

no port *slot***/***port*

Cisco AS5300

port *controller***:D**

no port *controller-***:D**

Cisco AS5800

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

no port {*shelf***/***slot***/***port***:D** | *shelf***/***slot***/***parent***:***port***:D**}

Cisco uBR925 Series

port {*slot***/***subunit***/***port*}

no port {*slot***/***subunit***/***port*}

Syntax Description

Cisco 1750 Series and Cisco 3700 Series

slot	Slot in which the voice interface cards (VIC) is installed. Valid entries are from 0 to 2.
port	Voice port. Valid entries are 0 and 1.

Cisco 2600 and Cisco 3600 Series

slot-number	Slot number in which the VIC is installed. Valid entries are from 0 to 3.
subunit	Subunit on the VIC in which the voice port is located. Valid entries are 0 and 1.
port	Voice port number. Valid entries are 0 and 1.
slot	Slot in which the voice port adapter is installed. Valid entries are 0 and 3.
port	VIC location. Valid entries are 0 and 3.
dso-group	DS-0 group number. Each defined DS-0 group number is represented on a separate voice port. This allows you to define individual DS-0s on the digital T1/E1 card.

MC3810

*slot***/***port*	Slot and port numbers. Slot is the slot number in the Cisco router in which the VIC is installed. The only valid entry is 1. Port is the voice port number. Valid ranges are as follows: • Analog voice ports: from 1 to 6. • Digital T1: from 1 to 24. • Digital E1: from 1 to 15, and from 17 to 31.

Cisco AS5300

controller	T1 or E1 controller.
:D	D channel that is associated with ISDN PRI.

Cisco AS5800

*shelf***/***slot***/***port*	T1 or E1 controller on the T1 card. Valid entries are as follows: • *shelf*: 0 to 9999 • *slot:* 0 to 11. • *port:* 0 to 11.
*shelf***/***slot***/***parent***:***port*	T1 controller on the T3 card. Valid entries for the *shelf* argument are from 0 to 9999. Valid entries for the *slot* variable are from 0 to 11. Valid entries for the *port* argument are from 1 to 28. The value for the *parent* argument is always 0.
:D	D channel associated with ISDN PRI.

Cisco 7200 Series

slot	Router location in which the voice port adapter is installed. Valid entries are from 0 to 3.
port	VIC location. Valid entries are 0 and 1.
dso-group	DS-0 group number. Each defined DS-0 group number is represented on a separate voice port. This allows you to define individual DS-0s on the digital T1/E1 card.
slot	Slot number in which the VIC is installed. Valid entries are from 0 to 3.

subunit-	Subunit on the VIC in which the voice port is located. Valid entries are 0 and 1.
port	Voice port number. Valid entries are 0 and 1.

Cisco uBR925

slot/subunit/port	(Optional) Displays information for the analog voice port that you specify with the *slot/subunit/port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a VIC where the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1.

Defaults

No port is configured.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(3)T	This command was implemented on the Cisco 2600.
11.3 MA	This command was implemented on the Cisco MC3810.
12.0(3)T	This command was implemented on the Cisco AS5300.
12.0(4)T	This command was implemented on the Cisco uBR924.
12.0(7)T	This command was implemented on the Cisco AS5800.
12.2(8)T	This command was implemented on the Cisco 1750 and Cisco 3700 series.
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T. This command does not support the extended echo canceller (EC) feature on the Cisco AS5300 or the Cisco AS5800.

Usage Guidelines

Use this command for calls coming from a telephony interface to select an incoming dial peer and for calls coming from the VoIP network to match a port with the selected outgoing dial peer.

This command applies only to POTS peers.

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

Examples

The following example associates a Cisco 3600 series POTS dial peer 10 with voice port 1, which is located on subunit 0 and accessed through port 0:

```
dial-peer voice 10 pots
 port 1/0/0
```

The following example associates a Cisco MC3810 POTS dial peer 10 with voice port 0, which is located in slot 1:

```
dial-peer voice 10 pots
 port 1/0
```

The following example associates a Cisco AS5300 POTS dial peer 10 with voice port 0:D:

```
dial-peer voice 10 pots
 port 0:D
```

The following example associates a Cisco AS5800 POTS dial peer 10 with voice port 1/0/0:D (T1 card):

```
dial-peer voice 10 pots
 port 1/0/0:D
```

Related Commands

Command	Description
prefix	Specifies the prefix of the dialed digits for a dial peer.

prefix

To specify the prefix of the dialed digits for a dial peer, use the **prefix** command in dial-peer configuration mode. To disable the prefix, use the **no** form of this command.

prefix *string*

no prefix

Syntax Description

string	Prefix of the telephone number that is associated with the specified dial peer. Valid numbers are 0 through 9 and a comma (,) to include a pause in the prefix.

Defaults

Null string

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for store-and-forward fax.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.
12.2(4)T	This command was implemented on the Cisco 1750.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.2(13)T	This command implemented on the Cisco 2600XM, Cisco ICS7750, and Cisco VG200.

Usage Guidelines

When an outgoing call is initiated to this dial peer, the **prefix** *string* value is sent to the telephony interface first, before the telephone number associated with the dial peer.

To configure different prefixes for dialed numbers on the same interface, you must configure different dial peers.

This command is applicable only to POTS dial peers. It applies to off-ramp store-and-forward fax functions.

Examples

The following example shows partial output from the **show running config** command, which shows that a prefix of 9 and a pause have been configured:

```
dial-peer voice 10 pots
 prefix 9,
.
.
.
```

Related Commands

Command	Description
answer-address	Specifies the full E.164 telephone number to be used to identify the dial peer of an incoming call.
destination-pattern	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.

show call active

To display active call information for voice calls or fax transmissions in progress, use the **show call active** command in user EXEC or privileged EXEC mode.

show call active {**fax** | **voice**} {[**brief** [**id** *identifier*] | **compact** [**duration** {**less** *time* | **more** *time*}] | **echo-canceller** | **id** *identifier*]}

Syntax Description

fax	Displays active store-and-forward fax calls.
voice	Displays active voice calls.
echo-canceller	Displays information about the state of the extended echo canceller (EC). You need to know in advance the hex ID to query the echo state. To find the hex ID, use the **show call active voice brief** command or the **show voice call status** command.
brief	(Optional) Displays a truncated version.
compact	(Optional) Displays a compact version.
duration	(Optional) Displays active calls that are longer or shorter than a specified time. Arguments and keywords are as follows: • **less**—Displays calls shorter than *time*. • **more**—Displays calls longer than *time*. • *time*—Elapsed time, in seconds. Range is from 1 to 2147483647. There is no default.
id *identifier*	Displays only the call with the specified *identifier*. Range is from 1 to FFFF.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 2600 series and Cisco 3600 series.
12.0(3)XG	This command was modified for Voice over Frame Relay (VoFR) applications.
12.0(4)XJ	This command was modified for store-and-forward fax on the Cisco AS5300.
12.0(4)T	This command was implemented on the Cisco 7200 series.
12.0(7)XK	This command was implemented on Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(2)E	This command was implemented on the Cisco 7500 series.
12.1(3)T	This command was modified for modem pass-through over VoIP on the Cisco AS5300.
12.1(5)XM	This command was implemented on the Cisco AS5800.
12.1(5)XM2	The command was implemented on the Cisco AS5350 and Cisco AS5400.

Release	Modification
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.
12.2(13)T	The **echo-canceller** keyword was added. The command output was modified with an extra reflector location when the extended EC is present; the largest reflector location is shown.

Usage Guidelines

Use this command to display the contents of the active call table. This command displays information about call times, dial peers, connections, quality of service, and other status and statistical information. The **voice** keyword displays information about all voice calls currently connected through the router or access server. When the extended EC is present, the **show call active voice** command displays the contents of the Ditech EC_CHAN_CTRL structure. Table 3 contains field name descriptions in the EC_CHAN_CTRL structure.

Table 3 EC_CHAN_CTRL Field Descriptions

Symbol	Field	Description
BYP0	Channel bypass	1 = Transparent bypass; EC is disabled. 0 = Cancel; EC is enabled.
TAIL3	Max tail	0 = 24 ms. 1 = 32 ms. 2 = 48 ms. 3 = 64 ms. **Note** This field should be set just higher than the anticipated worst round-trip tail delay.
REC3	Residual echo control	0 = Cancel only; echo is the result of linear processing; no nonlinear processing is applied. 1 = Suppress residual; residual echo is zeroed; simple nonlinear processing is applied (you might experience “dead ear” when talking). 2 = Reserved. 3 = Generate comfort noise (default).
FRZ0	h-register hold	1 = Freezes h-register; used for testing.
HZ0	h-register clear	Sending the channel command with this bit set clears the h-register.
TD3	Modem tone disable	0 = Ignore 2100-Hz modem answer tone. 1 = G.164 mode (bypass canceller if 2100-Hz tone). 2 = R. 3 = G.165 mode (bypass canceller for phase reversing tone only).

Table 3 EC_CHAN_CTRL Field Descriptions (continued)

Symbol	Field	Description
ERL0	Echo return loss	0 = 6 dB. 1 = 3 dB. 2 = 0 dB. 3 = R. Worst echo return loss (ERL) situation in which canceller still works.
HLC1	High level compensation	0 = No attenuation. 1 = 6 dB if clipped. On loud circuits, the received direction can be attenuated 6 dB if clipping is observed.
R0	Reserved	Must be set to 0 to ensure compatibility with future releases.

Examples

The following is sample output from the **show call active voice** command:

```
Router# show call active voice

Total call-legs:2

 GENERIC:
SetupTime=7587246 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=7587506
CallDuration=00:00:11
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=2
TransmitPackets=101
TransmitBytes=1991
ReceivePackets=550
ReceiveBytes=11000
VOIP:
ConnectionId[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
IncomingConnectionId[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
RemoteIPAddress=172.29.248.111
RemoteUDPPort=17394
RoundTripDelay=4 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=FALSE

SessionProtocol=cisco
SessionTarget=
OnTimeRvPlayout=10300
```

```
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=69 ms
ReceiveDelay=69 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
SignalingType=ext-signal
CallerName=
CallerIDBlocked=False
 GENERIC:
SetupTime=7587246 ms
Index=2
PeerAddress=133001
PeerSubAddress=
PeerId=133001
PeerIfIndex=8
LogicalIfIndex=7
ConnectTime=7587505
CallDuration=00:00:56
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=2801
TransmitBytes=56020
ReceivePackets=162
ReceiveBytes=3192
TELE:
ConnectionId=[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
IncomingConnectionId=[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
TxDuration=56030 ms
VoiceTxDuration=3210 ms
FaxTxDuration=0 ms
CoderTypeRate=g729r8
NoiseLevel=-44
ACOMLevel=-13
OutSignalLevel=-45
InSignalLevel=-45
InfoActivity=2
ERLLevel=7
EchoCancellerMaxReflector=64
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
```

Table 4 shows significant fields in the display.

Table 4 show call active voice Field Descriptions

Field	Description
ACOM Level	Current ACOM level for this call. This value is the sum of the echo return loss, echo return loss enhancement, and nonlinear processing loss for this call.
CallOrigin	Call origin: answer or originate.
CallState	Current state of the call.
CoderTypeRate	Negotiated coder transmit rate of voice or fax compression during this call.
ConnectionId	Global call identifier for this gateway call.
ConnectTime	Time at which the call was connected.
Dial-Peer	Tag of the dial peer that is transmitting this call.
EchoCancellerMaxReflector=64	The location of the largest reflector, in milliseconds. The reflector size does not exceed the configured echo path capacity. For example, if 32 ms is configured, the reflector does not report beyond 32 ms.
ERLLevel	Current echo return loss (ERL) level for this call.
FaxTxDuration	Duration of fax transmission from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithInterpolation	Duration of the voice signal played out with the signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithPrediction	Duration of the voice signal played out with the signal synthesized from parameters or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call. Examples of such pullout are frame-eraser or frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithRedundancy	Duration of the voice signal played out with the signal synthesized from redundancy parameters available because voice data was lost or not received in time from the voice gateway for this call.
GapFillWith Silence	Duration of the voice signal replaced with silence because voice data was lost or not received in time for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during this call.
Index	Dial-peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call.
InSignalLevel	Active input signal level from the telephony interface used by this call.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low-water-mark Voice Playout FIFO Delay during this call.
NoiseLevel	Active noise level for this call.
OnTimeRvPlayout	Duration of the voice playout from data received in time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.

Table 4 show call active voice Field Descriptions (continued)

Field	Description
OutSignalLevel	Active output signal level to telephony interface used by this call.
PeerAddress	Destination pattern associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice-port index number for this peer.
PeerSubaddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the Decoder Delay during this call.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system User Datagram Protocol (UDP) listener port to which voice packets are transmitted.
RoundTripDelay	Voice packet round-trip delay between the local and remote system on the IP backbone during this call.
SelectedQoS	Selected Resource Reservation Protocol. Protocol (RSVP) quality of service (QoS) for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote router via the IP backbone.
SessionTarget	Session target of the peer used for this call.
SetupTime	Value of the system UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted from this peer during this call.
TransmitPackets	Number of packets transmitted from this peer during this call.
TxDuration	Duration of transmit path open from this peer to the voice gateway for this call.
VADEnable	Whether voice activity detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to the voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

The following is an example of the **show call active voice** command used with the extended echo canceller. The number 9 represents the hexadecimal ID of an active voice call.

```
Router# show call active voice echo-canceller 9

ACOM=-65  ERL=45
Echo canceller control words=6C 0
Bypass=OFF  Tail=64  Residual ecan=Comfort noise
Freeze=OFF  Modem tone disable=Ignore 2100Hz tone
Worst ERL=6  High level compensation=OFF
Max amplitude reflector (in msec)=5
Ecan version = 8180
```

The following is sample output from the **show call active voice brief** command:

```
Router# show call active voice brief
```

```
<ID>:<start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
  dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
  MODEMPASS <method> buf:<fills>/<drains> loss <overall%>
<multipkt>/<corrected>
   last <buf event time>s dur:<Min>/<Max>s
 FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
 ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
 Tele <int>:tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l>
dBm
  MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent>
total:<rcvd>/<sent>/<drops>
 Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt:
<type>/<manf>
 bw:<req>/<act> codec:<audio>/<video>
  tx:<audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120
pkts>/<t120 bytes>
 rx:<audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120
pkts>/<t120 bytes>


Total call-legs:2
1269 :7587246hs.1 +260 pid:0 Answer  active
 dur 00:07:14 tx:590/11550 rx:21721/434420
IP 172.29.248.111:17394 rtt:3ms pl:431850/0ms lost:0/0/0 dela
y:69/69/70ms g729r8

1269 :7587246hs.2 +259 pid:133001 Originate 133001 active
 dur 00:07:14 tx:21717/434340 rx:590/11550
 Tele 1/0:1 (2):tx:434350/11640/0ms g729r8 noise:-44 acom:-19
i/0:-45/-45 dBm
```

The following is sample output from the **show call active voice echo-canceller** command.

```
Router# show call active voice echo-canceller 10

ACOM=-15  ERL=7
Echo canceller control words=6C 0
Bypass=OFF  Tail=64  Residual ecan=Comfort noise
Freeze=OFF  Modem tone disable=Ignore 2100Hz tone
Worst ERL=6  High level compensation=OFF
Max amplitude reflector (in msec)=64
Router#
```

The call ID number (10 in the example above) changes with every new active call. When an active call is up, you must enter the **show call active voice brief** command to obtain the call ID number. The call ID must be converted to hex if you want to use the **show call active voice echo-canceller x** command (**x** = call ID converted to hex).

The following are call ID examples converted to hex (generally increment by 2):

Decimal	Hex
2	2
4	4
6	6

Decimal	Hex
8	8
10	A
12	C

Alternatively, you can use the **show voice call status** command to obtain the call ID. The call ID output is already in hex form when you use this command:

```
Router# show voice call status

CallID      CID  ccVdb      Port      DSP/Ch  Called #   Codec    Dial-peers
0x1         11CE 0x02407B20 1:0.1     1/1     1000       g711ulaw 2000/1000
```

Related Commands

Command	Description
show call history	Displays the call history table.
show dial-peer voice	Displays configuration information and call statistics for dial peers.
show frame-relay pvc	Displays statistics for PVCs associated with Frame Relay interfaces.
show frame-relay vofr	Displays information about the FRF.11 subchannels associated with VoFR DLCIs.
show num-exp	Displays how number expansions are configured in VoIP.
show voice call status	Displays the call status for voice ports on Cisco router or concentrator.
show voice-port	Displays configuration information about a specific voice port.

show voice call

To show the call status for voice ports on a Cisco router or concentrator, use the **show voice call** command in EXEC mode.

Cisco 827, Cisco 1700 Series, Cisco 7750, and Cisco MC3810 with Analog Voice Ports

show voice call [*slot*/*port* | **status** *call-id* [**sample** *sample-period*] | **summary**]]

Cisco 2600, Cisco 3600, Cisco 3700 Series, Cisco CVA122, Cisco uBR925, and Cisco VG200 with Analog Voice Ports

show voice call [*slot*/*subunit*/*port* | **status** *call-id* [**sample** *sample-period*] | **summary**]]

Cisco 2600, Cisco 3600, and Cisco 3700 Series with Digital Voice Ports (with T1 Packet Voice Trunk Network Modules)

show voice call [*slot*/*port***:***ds0-group* | **status** *call-id* [**sample** *sample-period*] | **summary**]]

Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, Cisco AS5850, Cisco 7200 Series, Cisco 7500 Series, and Cisco VG200 with Digital Voice Ports with Digital Voice Ports

show voice call [*slot*/*port*:*ds0-group* | **status** *call-id* [**sample** *sample-period*] | **summary**]]

Cisco MC3810 with Digital Voice Ports

show voice call [*slot***:***ds0-group* | **status** *call-id* [**sample** *sample-period*] | **summary**]]

Syntax Description

Cisco 827, Cisco 1700 Series, Cisco ICS7750, and Cisco MC3810 with Analog Voice Ports

:

slot/*port*	(Optional) Displays information for the analog voice port that you specify with the *slot*/*port* designation. • The *slot* argument is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. • The *port* argument specifies an analog voice port number. Valid entries are from 1 to 6. • If you are running the Cisco ICS7750 with an ASI81 (8-port analog card), the number of analog voice ports can range from 0 to 8. Valid entries depend on the type of analog card you have. For example, if you have a 2-port Foreign Exchange Station (FXS), the number of analog voice ports can be 0 or 1.
status	(Optional) Displays status information of all voice ports.
call-id	(Optional) A call ID.
sample *sample-period*	(Optional) Show status over this sampling interval. The *sample-period* is the amount of time, in seconds. Range is from 1 to 30. Default is 10.
summary	(Optional) Displays a summary of all voice ports.

Cisco 2600 series, Cisco 3600 Series, Cisco 3700 Series, Cisco CVA122, Cisco uBR925, and Cisco VG200 with Analog Voice Ports

slot/*subunit*/*port*	(Optional) Displays information for the analog voice port that you specify with the *slot*/*subunit*/*port* designation. • The *slot* argument specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • The *subunit* argument specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) • The *port* argument specifies an analog voice port number. Valid entries are 0 and 1.
status	(Optional) Displays status information of all voice ports.
call-id	(Optional) A call ID.
sample *sample-period*	(Optional) Show status over this sampling interval. The *sample-period* is the amount of time in seconds. Range is from 1 to 30. Default is 10.
summary	(Optional) Displays a summary of all voice ports.

Cisco 2600, Cisco 3600, and Cisco 3700 Series with Digital Voice Ports (with T1 Packet Voice Trunk Network Modules)

slot/*port*:*ds0-group*	(Optional) Displays information for the digital voice port that you specify with the *slot*/*port*:*ds0-group* designation. • The *slot* argument specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • The *port* argument specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. (One VWIC fits in an NM.) • The *ds0-group* argument specifies a T1 or E1 logical port number. Valid entries are from 0 to 23 for T1 and from 0 to 30 for E1.
status	(Optional) Displays status information of all voice ports.
call-id	(Optional) A call ID.
sample *sample-period*	(Optional) Show status over this sampling interval. The *sample-period* is the amount of time in seconds. Range is from 1 to 30. Default is 10.
summary	(Optional) Displays a summary of all voice ports.

Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, Cisco AS5850, Cisco 7200 Series, Cisco 7500 Series, and Cisco VG200 with Digital Voice Ports with Digital Voice Ports

slot/*port*:*ds0-group*	(Optional) Displays information for the digital voice port that you specify with the *slot*/*port*:*ds0-group* designation. • The *slot* argument specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • The *port* argument specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. (One VWIC fits in an NM.) • The *ds0-group* argument specifies a T1 or E1 logical port number. Valid entries are from 0 to 23 for T1 and from 0 to 30 for E1.
status	(Optional) Displays status information of all voice ports.
call-id	(Optional) A call ID.
sample *sample-period*	(Optional) Show status over this sampling interval. The *sample-period* is the amount of time in seconds. Range is from 1 to 30. Default is 10.
summary	(Optional) Displays a summary of all voice ports.

Cisco MC3810 with Digital Voice Ports

:

slot/*port*:*ds0-group*	(Optional) Displays information for the digital voice port that you specify with the *slot*/*port*:*ds0-group* designation. • The *slot* argument specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • The *port* argument specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. (One VWIC fits in an NM.) • The *ds0-group* argument specifies a T1 or E1 logical port number. Valid entries are from 0 to 23 for T1 and from 0 to 30 for E1.
status	(Optional) Displays status information for all voice ports.
call-id	(Optional) A call ID.
sample *sample-period*	(Optional) Show status over this sampling interval. The *sample-period* is the amount of time in seconds. Range is from 1 to 30. Default is 10.
summary	(Optional) Displays a summary for all voice ports.

Command Modes

User EXEC or privileged EXEC

Command History

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

Release	Modification
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(13)T	This command was modified with the **status**, *call-id*, and **sample** keywords and arguments. This command is available on all voice platforms.

Usage Guidelines

The following platforms do not support The Enhanced ITU-T G.168 Echo Cancellation feature in Cisco IOS Release 12.2(13)T: Cisco 827, Cisco AS5x00, Cisco CVA122, Cisco uBR925

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over IP.

This command shows call-processing and protocol state-machine information for a voice port, if it is available. It also shows information on the DSP channel associated with the voice port, if it is available. All real-time information in the DSP channel, such as jitter and buffer overrun, is queried to the DSP channel, and asynchronous responses are returned to the host side.

If no call is active on a voice port, the **show voice call summary** command displays only the VPM (shutdown) state. If a call is active on a voice port, the VTSPS state is shown. For an on-net call or a local call without local-bypass (not cross-connected), the CODEC and VAD fields are displayed. For an off-net call or a local call with local-bypass, the CODEC and VAD fields are not displayed.

This command does not show CODEC and VAD because this information is in the summary display.

This command provides the status at the following levels of the call handling module:

- Tandem switch
- End-to-end call manager
- Call processing state machine
- Protocol state machine

If you use the **show voice call status** command by itself, an immediate list of all the active calls is shown. You can use the *call-id* argument to request that the digital signal processor (DSP) associated with the *call-id* be queried for run-time statistics twice, once immediately and a second time after **sample** *sample-period* seconds. You can find the value of the hexadecimal ID by using the **show voice call status** command.

On a router that supports large numbers of active calls, you can use the | (pipe) option. The following keywords can be used to reduce and select the output:

Keyword	Description
append	Appends redirected output to URL (URLs supporting append operation only). Can be used with the following: The **append** keyword can be used with the following: • **flash**—Uniform Resource Locator • **ftp**—Uniform Resource Locator • **nvram**—Uniform Resource Locator • **pram**—Uniform Resource Locator • **rcp**—Uniform Resource Locator • **slot0**—Uniform Resource Locator • **slot1**—Uniform Resource Locator • **tftp**—Uniform Resource Locator
begin	Begins with the line that matches. Can be used with the following: • **line**—Regular expression
exclude	Excludes lines that match. Can be used with the following: • **line**—Regular expression
include	Includes lines that match. Can be used with the following: • **line**—Regular expression
redirect	Redirects output to URL. Can be used with the following: • **flash**—Uniform Resource Locator • **ftp**—Uniform Resource Locator • **nvram**—Uniform Resource Locator • **pram**—Uniform Resource Locator • **rcp**—Uniform Resource Locator • **slot0**—Uniform Resource Locator • **slot1**—Uniform Resource Locator • **tftp**—Uniform Resource Locator
tee	Copies output to URL. Can be used with the following: • **flash**—Uniform Resource Locator • **ftp**—Uniform Resource Locator • **nvram**—Uniform Resource Locator • **pram**—Uniform Resource Locator • **rcp**—Uniform Resource Locator • **slot0**—Uniform Resource Locator • **slot1**—Uniform Resource Locator • **tftp**—Uniform Resource Locator

Examples

The following is sample output from the **show voice call summary** command for voice ports on a Cisco MC3810, showing two local calls connected without local bypass:

```
Router# show voice call summary

PORT    CODEC    VAD VTSP STATE            VPM STATE
======= ======== === ===================== ========================
0:17.18                                     *shutdown*
0:18.19 g729ar8   n  S_CONNECT             FXOLS_OFFHOOK
0:19.20                                    FXOLS_ONHOOK
0:20.21                                    FXOLS_ONHOOK
0:21.22                                    FXOLS_ONHOOK
0:22.23                                    FXOLS_ONHOOK
0:23.24                                    EM_ONHOOK
1/1                                        FXSLS_ONHOOK
1/2                                        FXSLS_ONHOOK
1/3                                        EM_ONHOOK
1/4                                        EM_ONHOOK
1/5                                        FXOLS_ONHOOK
1/6     g729ar8   n  S_CONNECT             FXOLS_CONNECT
```

The following is sample display from the **show voice call summary** command for voice ports on a Cisco MC3810, showing two local calls connected with local bypass:

```
Router# show voice call summary

PORT    CODEC    VAD VTSP STATE            VPM STATE
======= ======== === ===================== ========================
0:17.18                                     *shutdown*
0:18.19              S_CONNECT             FXOLS_OFFHOOK
0:19.20                                    FXOLS_ONHOOK
0:20.21                                    FXOLS_ONHOOK
0:21.22                                    FXOLS_ONHOOK
0:22.23                                    FXOLS_ONHOOK
0:23.24                                    EM_ONHOOK
1/1                                        FXSLS_ONHOOK
1/2                                        FXSLS_ONHOOK
1/3                                        EM_ONHOOK
1/4                                        EM_ONHOOK
1/5                                        FXOLS_ONHOOK
1/6                  S_CONNECT             FXOLS_CONNECT
```

The following sample output from the **show voice call** command for analog voice ports on a Cisco MC3810:

```
Router# show voice call

1/1 vpm level 1 state = FXSLS_ONHOOK
vpm level 0 state = S_UP
1/2 vpm level 1 state = FXSLS_ONHOOK
vpm level 0 state = S_UP
1/3 is shutdown
1/4 vtsp level 0 state = S_CONNECT
vpm level 1 state = S_TRUNKED
vpm level 0 state = S_UP
1/5 vpm level 1 state = EM_ONHOOK
vpm level 0 state = S_UP
1/6 vpm level 1 state = EM_ONHOOK
vpm level 0 state = S_UP
sys252#show voice call 1/4
1/4 vtsp level 0 state = S_CONNECT
vpm level 1 state = S_TRUNKED
vpm level 0 state = S_UP
router#    ***DSP VOICE VP_DELAY STATISTICS***
```

```
Clk Offset(ms): 1445779863, Rx Delay Est(ms): 95
Rx Delay Lo Water Mark(ms): 95, Rx Delay Hi Water Mark(ms): 125
        ***DSP VOICE VP_ERROR STATISTICS***
Predict Conceal(ms): 10, Interpolate Conceal(ms): 0
Silence Conceal(ms): 0, Retroact Mem Update(ms): 0
Buf Overflow Discard(ms): 20, Talkspurt Endpoint Detect Err: 0
        ***DSP VOICE RX STATISTICS***
Rx Vox/Fax Pkts: 537, Rx Signal Pkts: 0, Rx Comfort Pkts: 0
Rx Dur(ms): 50304730, Rx Vox Dur(ms): 16090, Rx Fax Dur(ms): 0
Rx Non-seq Pkts: 0, Rx Bad Hdr Pkts: 0
Rx Early Pkts: 0, Rx Late Pkts: 0
        ***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 567, Tx Sig Pkts: 0, Tx Comfort Pkts: 0
Tx Dur(ms): 50304730, Tx Vox Dur(ms): 17010, Tx Fax Dur(ms): 0
        ***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0
        ***DSP LEVELS***
TDM Bus Levels(dBm0): Rx -70.3 from PBX/Phone, Tx -68.0 to PBX/Phone
TDM ACOM Levels(dBm0): +2.0, TDM ERL Level(dBm0): +5.6
TDM Bgd Levels(dBm0): -71.4, with activity being voice
```

The following is sample output from the **show voice call** command for analog voice ports on a Cisco 7200, which shows the DSPfarm, T1 interface, and DS-0 or TLM slot configuration:

```
Router# show voice call 6/0:0

6/0:0 1 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 2 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 3 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 4 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 5 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 6 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 7 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 8 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 9 -         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 10-         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 11-         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
6/0:0 12-         -  -                    vpm level 1 state = FXOGS_ONHOOK
vpm level 0 state = S_UP
```

The following is sample output of the **show voice call status** command on a Cisco 2600 series. You can use this command to obtain the call ID rather than the **show call active brief** command; the call ID output of the **show voice call status** command is already in hex form.

```
Router# show voice call status

CallID     CID  ccVdb      Port      DSP/Ch  Called #   Codec    Dial-peers
0x1        11CE 0x02407B20 1:0.1     1/1     1000       g711ulaw 2000/1000
1 active call found
```

Using the *call-id* argument with the **status** keyword is a generic means to identify active calls. If the *call-id* is omitted, the enquiry shows all active voice calls. The following example shows a list of all active calls with relevant identifying information is shown:

```
Router# show voice call status

CallID    CID    ccVdb       Port     DSP/Ch  Called #    Codec     Dial-peers
0x3       11D4   0x62972834  1/0/0    1/1      10001      g711ulaw 1/2
0x4       11D4   0x62973AD0  1/0/1    2/1     *10001      g711ulaw 2/1
0xA       11DB   0x62FE9D68  1/1/0    3/1     *2692       g729r8   0/2692
2 active calls found
```

Table 5 shows output field descriptions for the **show voice call** command using the **status** keyword.

Table 5 show voice call status Field Descriptions

Field	Description
CallID	Hexidecimal number that is used for further enquiry. It is the monotonically increasing number that call control maintains for each call leg (ccCallID_t).
CID	Conglomerate value derived from the GUID that appears in the **show call active brief** command.
ccVdb	Value that is displayed in many other debugs to identify these call legs.
Port	Voice port.
DSP/Ch	DSP and channel allocated to this call leg. The format of these values is platform dependent (particularly the Cisco AS5300, which shows the DSP number as a 3-digit number, <VFC#><DSPM#><DSP#>). Time-slot information is also in the output for digital ports. For example, if you are using a digital port, the time slot is also returned: dsp/ch/time slot.
Called #	Called number. 10001—No '*' denotes a call leg that originates a call to the Called #. Two of the call legs in the example constitute one locally switched call and one network call; hence they refer to two active calls. * 10001—The '*' in front of the second number in the Called # column denotes that this is a destination call leg (for example, this number was called with Called #).
Codec	Codec type.
Dial-peers	Dial-peer.

Note Only one call may be queried at a time. If you attempt queries from different ports (console and Telnet), and if a query is in progress on another port, the system requests that you wait for completion of that query. You can query any call from anywhere at anytime except during the sample interval for an enquiry already in progress. This simplifies the implementation significantly and does not reduce the usefulness of the command.

The following example shows echo return loss (ERL) reflector information, where 3 is the hexadecimal id and the sample period is 10 seconds:

```
Router# show voice call status 3 sample 10

Gathering information (10 seconds)...
CallID    Port     DSP/Ch  Codec    Rx/Tx     ERL         Jitter
0x3       1/0/0    1/1     g711ulaw 742/154   5.6         50/15
```

Table 6 show voice call status Field Descriptions

Field	Description
CallID	Hexidecimal number that is used for further enquiry. It is the monotonically increasing number that call control maintains for each call leg (ccCallID_t).
CID	Conglomerate value derived from the GUID that appears in the **show call active brief** command.
Port	Voice port.
DSP/Ch	DSP and channel allocated to this call leg. The format of these values is platform dependent (particularly the Cisco AS5300, which shows the DSP number as a 3-digit number, <VFC#><DSPM#><DSP#>). Time-slot information is also in the output for digital ports. For example, if you are using a digital port, the time slot is also returned: dsp/ch/time slot.
Codec	Codec type.
Rx/Tx	Receive and transmit.
ERL	Measure of the ERL (in dB) as reported by the DSP.
Jitter	Value of the delay and the jitter of the packets around that delay.

If the router is running the extended echo canceller, output looks similar to the following when the command above is used. The output shows a new value under ERL/Reflctr: the time difference, in milliseconds, between the original signal and the loudest echo (peak reflector) as detected by the echo canceller.

```
Gathering information (10 seconds)...
CallID    Port     DSP/Ch  Codec    Rx/Tx     ERL/Reflctr Jitter
0x3       1/0/0    1/1     g711ulaw 742/154   5.6/12      50/15
```

The following sample shows the NextPort version of the standard echo canceller. (Time-slot information is also in the output for digital ports.)

```
Router# show voice call status

CallID     CID  ccVdb      Port       DSP/Ch  Called #   Codec    Dial-peers
0x97       12BB 0x641B0F68 3/0:D.1    1012/2  31001      g711ulaw 3/31000
0x99       12BE 0x641B0F68 3/0:D.2    1012/3  31002      g711ulaw 3/31000
2 active calls found
```

The following is sample output on a Cisco 827:

```
Router# show voice call status 23 sample 20

Gathering information (20 seconds)...
CallID     Port      DSP/Ch  Codec    Rx/Tx     ERL         Jitter
0x23       1         0/1     g729br8  73/4      24.6        58/12
```

When using the **test call id** command, you must specify a call ID. You can obtain the call ID by using the **show voice call status** command. The first parameter displayed in the output shows the call ID. The hexadecimal call ID is highlighted in the example:

```
Router# show voice call status

CallID     CID  ccVdb       Port      DSP/Ch  Called #   Codec    Dial-peers
0x2        11D1 0x62FE6478 1/0/0      1/1     10001      g711ulaw 1/2
0x3        11D1 0x62FE80F0 1/0/1      2/1    *10001      g711ulaw 2/1
1 active call found
```

Note Do not use the 0x prefix in the *call-id* argument when you enter the resulting call ID in the **test call status** command.

When a call terminates during the specified sample period, the following output message is returned:

```
CallID call id cannot be queried
CallID call id second sample responses unavailable
```

The following example shows keyword choices when using the **show voice call** command with the **|** (pipe) option:

```
Router# show voice call | ?

  append    Append redirected output to URL (URLs supporting append operation
            only)
  begin     Begin with the line that matches
  exclude   Exclude lines that match
  include   Include lines that match
  redirect  Redirect output to a URL
  tee       Copy output to a URL
```

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all VoIP and POTS dial peers configured on the router.
show voice dsp	Displays the current status of all DSP voice channels.
show voice port	Displays configuration information about a specific voice port.
test call id	Manipulates the echo canceller and jitter buffer parameters in real time.

test call id

To test mode settings to allow manual manipulation of the echo canceller b-register for G.168-like tests, use the **test call id** command in privileged EXEC mode.

test call id *call-id* {**echo-canceller** {**coverage** *range-in-ms* | **erl worst-case** {**0** | **3** | **6**}| **h-register** {**clear** | **freeze** | **thaw**}} | **playout-delay** {**fixed** | **adaptive** {*nominal-delay min-delay max-delay*}}}

Syntax Description

call-id	Hexadecimal ID of an active voice call. Range is from 0 to FFFFFFFF.
echo-canceller	Tests the echo canceller on an active voice call.
coverage *range-in-ms*	Echo canceller coverage in milliseconds. Valid values are 0, 8, 16, 24, 32, 48, 64, and 128. Default values are as follows: • Standard echo canceller (Cisco-proprietary G.165 EC)—8 ms • Extended echo canceller—64 • NextPort firmware—8 See the "Usage Guidelines" section for more information about default values.
erl worst-case {**0** | **3** | **6**}	Worst-case Echo Return Loss (ERL), in decibels (dB). Valid values are 0, 3, or 6. Default is 6. **Note** The **echo-canceller erl worst-case** keywords combine to form a tunable parameter available with the extended echo canceller only. The **erl** option is available only with the extended echo canceller.
h-register	Controls the extended echo canceller h-register.
{**clear** | **freeze** | **thaw**}	Clears, freezes, or thaws a call in the extended echo canceller h-register.
playout-delay	Resets the playout buffering on the associated digital signal processors (DSPs) to the requested values. If **fixed** *fixed-delay* is selected, the first parameter only is required and used. If all three parameters are selected, they are accepted, but the last two are ignored. If **adaptive** *nominal-delay min-delay max-delay* is selected, all three values are required and used.
fixed *fixed-delay*	Tests the fixed playout-delay mode. Jitter buffer size does not adjust during a call; a constant playout delay is added. The *fixed-delay* argument is nominal delay in ms. Range is from 0 to 1500.
adaptive *nominal-delay min-delay max-delay*	Tests the adaptive playout-delay mode. Adjusts jitter buffer size and amount of playout delay during a call on the basis of current network conditions. If the **adaptive** keyword is used, *nominal-delay*, *min-delay*, and *max-delay* are sanity checked for maximum delay being greater than or equal to the nominal delay, which is greater than or equal to the minimum delay. Nominal delay range is from 0 to 1500 ms. Minimum delay range is from 10 to 80 ms. Maximum delay range is from 40 to 1700 ms. **Note** These options cause audible disturbance to the call and should be used with care.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(13)T	This command was introduced on all voice platforms with echo cancellation and extended echo cancellation.

Usage Guidelines

To obtain the *call-id* argument, use the **show voice call status** command, as shown in the following is an example. The first parameter in the output is the call ID.

Note Do not use the "0x" prefix in the *call-id* argument when you enter the resulting call ID in the **show voice call status** command.

```
Router# show voice call status

CallID      CID  ccVdb        Port      DSP/Ch  Called #   Codec    Dial-peers
0x2         11D1 0x62FE6478 1/0/0      1/1     10001      g711ulaw 1/2
0x3         11D1 0x62FE80F0 1/0/1      2/1    *10001      g711ulaw 2/1
1 active call found
```

Some of the options in the Syntax Description table can be used only on specific platforms that are running the extended echo canceller. Table 7 lists the platforms supported with this feature and whether the standard (TI C54x voice-based platforms) or the extended (NextPort/Conexant voice-based platforms) echo canceller is available on that platform. A disabled state is indicated by 0.

Table 7 Echo Canceller Types and Canceller Coverage Ranges

Platform	Echo Canceller Type	Echo Canceller Coverage Range
Cisco 1700 series	Standard	0, 8, 16, 24, 32
Cisco 2400 series	Standard	0, 8, 16, 24, 32
Cisco 2600 series	Standard	0, 8, 16, 24, 32
	Extended	0, 24, 32, 48, 64
Cisco 3600 series	Standard	0, 8, 16, 24, 32
	Extended	0, 24, 32, 48, 64
Cisco 7200	Standard	0, 8, 16, 24, 32
	Extended	0, 24, 32, 48, 64
Cisco 7750	Standard	0, 8, 16, 24, 32
Cisco 827	Standard	0, 8, 16, 24, 32
Cisco AS5300	Standard	0, 8, 16, 24, 32
	Extended	0, 24, 32, 48, 64
Cisco AS5350	NextPort	0, 8, 16, 24, 32, 64, 128
Cisco AS5400	NextPort	0, 8, 16, 24, 32, 64, 128
Cisco AS5800	Standard	0, 8, 16, 24, 32
Cisco AS5850	NextPort	0, 8, 16, 24, 32, 64, 128
Cisco CVA122	Standard	0, 8, 16, 24, 32

Table 7 Echo Canceller Types and Canceller Coverage Ranges (continued)

Platform	Echo Canceller Type	Echo Canceller Coverage Range
Cisco MC3810	Standard	0, 8, 16, 24, 32
	Extended	0, 24, 32, 48, 64
Cisco uBR925	Standard	0, 8, 16, 24, 32

Note The keywords and arguments in the Syntax Description table requests that the specified parameters be sent to the DSP using the normal DSP control message mechanism expecting an immediate effect. You can expect a short discontinuity and settling period for the voice stream. These parameters have effect only for the duration of the call. Echo-canceller and playout parameters revert to the values defined in the configuration on the next call using that DSP.

You can use this command with the extended echo canceller, which allows you to configure the voice card in a router individually, or with the standard echo canceller, in which the configuration occurs implicitly on the router. The following two new output messages are possible with the extended echo cancellation feature when either an extended-only or a standard-only echo cancellation function is requested:

```
Extended echo canceller not active for CallID callID
Basic echo canceller not active for CallID callID
```

The CLI help strings typically show which version of echo canceller is running and if it is valid for the requested function. For example:

```
Router# test call id 3 echo-canceller erl worst-case ?

  0   worst case extended echo canceller operation at 0 dB ERL
  3   worst case extended echo canceller operation at 3 dB ERL
  6   worst case extended echo canceller operation at 6 dB ER

Router# test call id 3 echo-canceller coverage ?

  0    disable echo-canceller
  16   16 ms echo canceller coverage (basic only)
  24   24 ms echo canceller coverage (basic & extended)
  32   32 ms echo canceller coverage (basic & extended)
  48   48 ms echo canceller coverage (extended only)
  64   64 ms echo canceller coverage (extended only)
  8    8 ms  echo canceller coverage (basic only)
```

In its section on testing echo cancellers, ITU-T specification G.168 invents a hypothetical device in the EC called an h-register. The h-register stores the impulse response of the echo path and invents actions such as “clear the h-register,” “contents of the h-register are frozen,” and “thaw” to undo the “freeze.” The h-register is the filter within EC used to estimate the echo. If it freezes, its filter coefficients do not adapt to the signal. If there is a significant change in the signal characteristic, such as power level or delay, echo is heard.

The h-register test mode settings allow manual manipulation of the EC h-register for G.168-like tests. Actual G.168 testing is embedded in the digital signal processor (DSP) and does not require explicit Cisco IOS control of the h-register. The call ID must be a valid active telephony call leg ID as displayed by entering the **show call active brief** command in privileged EXEC mode.

Refer to the *Extended ITU-T G.168 Echo Cancellation* feature module for more information about the extended echo canceller.

Examples

The following example experiments in real time with the parameters of an active call. In this example, the nominal delay for both the **adaptive** and **fixed** options is 5 ms; the minimum delay for the **adaptive** option is 10 ms; and the maximum delay for the **adaptive** option is 40 ms.

```
Router# test call id 99 playout-delay fixed 5
Router# test call id 99 playout-delay adaptive 5 10 40
```

The *call-id* argument is a generic means to identify active calls. The **playout-delay** keyword resets the playout buffering on the associated DSPs to the requested values. If the **fixed** mode is selected, there is only one fixed delay parameter. If the **adaptive** mode is selected, all three values are required and used.

If the fixed mode is selected, **fixed** *fixed-delay* is range-checked at 0 through 1500. If the **adaptive** mode is selected, the three argument values are sanity checked for maximum delay is greater than or equal to nominal delay, which is greater than or equal to the minimum delay. Options for the **adaptive** keyword are as follows:

nominal-delay—Range-checked at 0 to 1500

minimum-delay—Range-checked at 10 to 80

maximum-delay—Range checked at 40 to 1700

Note These options cause audible disturbance to the call. Use them carefully.

The following example sets the fixed delay to 0, which is the minimum value allowed:

```
Router# test call id 99 playout-delay fixed 0
```

The following example sets the minimum delay, nominal delay, and maximum delay. The maximum value allowed for each parameter is implemented:

```
Router# test call id 99 playout-delay adaptive 80 1500 1700
```

The following example tests the echo canceller on an active voice call on a Cisco AS5350 using the NextPort version of the standard echo canceller and a call ID value of 99:

```
Router# test call id 99 echo-canceller
```

The following example tests the playout delay parameters on an active voice call on a Cisco AS5350 using the NextPort version of the standard echo canceller and a call ID value of 99:

```
Router# test call id 99 playout-delay
```

The following example tests echo canceller coverage using a call ID value of 99:

```
Router# test call id 99 echo-canceller coverage
```

The following example tests extended echo canceller ERL parameters using a call ID value of 99:

```
outer# test call id 99 echo-canceller erl
```

The following example controls the extended echo canceller H-register using a call ID value of 99:

```
Router# test call id 99 echo-canceller h-register
```

The **echo-canceller coverage** keywords reset the echo canceller range on the associated DSPs to the new value, where 0 is the equivalent of switching the echo canceller off. Each value in the list shows whether it is supported on the basic or the extended echo canceller.

```
Router# test call id 99 echo-canceller coverage ?

  0   disable echo-canceller
  16  16 ms echo canceller coverage (basic only)
```

```
  24   24 ms echo canceller coverage (basic & extended)
  32   32 ms echo canceller coverage (basic & extended)
  48   48 ms echo canceller coverage (extended only)
  64   64 ms echo canceller coverage (extended only)
  8    8  ms echo canceller coverage (basic only)
```

The **erl worst-case** [*0* | *3* | *6*] syntax reflects the new tunable argument available with the extended echo canceller only. The following example uses a worst-case erl value of 3 dB:

```
Router# test call id 99 echo-canceller erl test extended echo canceller worst-case erl 3
```

The following is sample output from the **test call** command in privileged EXEC mode using a value of 02 for the call ID argument:

```
Router# test call ID 02 echo-canceller h-register ?

  clear   Clear call echo canceller h register
  freeze  Freeze call echo canceller h register
  thaw    Thaw call echo canceller h register
```

The **echo-canceller coverage** keywords reset the echo canceller range on the associated DSPs to the new value, where 0 is the equivalent of switching the echo canceller off. Each value in the list shows whether it is supported on the basic or the extended echo canceller.

```
Router# test call id 99 echo-canceller coverage ?

  0    disable echo-canceller
  16   16 ms echo canceller coverage (basic only)
  24   24 ms echo canceller coverage (basic & extended)
  32   32 ms echo canceller coverage (basic & extended)
  48   48 ms echo canceller coverage (extended only)
  64   64 ms echo canceller coverage (extended only)
  8    8  ms echo canceller coverage (basic only)
```

The **erl worst-case** [*0* | *3* | *6*] syntax reflects the new tunable argument available with the extended echo canceller only. The following example tests the extended echo canceller operation worst-case ERL at 3 dB:

```
Router# test call id 99 echo-canceller erl worst-case 3
```

The following example clears a call on the echo canceller h register using a value of 02 for the call ID argument:

```
Router# test call ID 02 echo-canceller h-register clear
```

Related Commands

Command	Description
echo cancel coverage	Enables the cancellation of voice that is sent out the interface and is received on the same interface.
show call active	Displays active call information for voice calls or fax transmissions.
show voice call status	Shows the real-time call status for voice ports.

voice-card

To enter voice-card configuration mode to configure resources on the network module, use the **voice-card** command in global configuration mode.

voice-card *slot*

Syntax Description

slot	Slot number for the card to be configured. The following platform-specific numbering schemes apply: • Cisco 2600 series: – 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. • Cisco MC3810 with one or two high-performance voice-compression modules (HCMs) installed: – 0 applies to the entire chassis.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XK	The command was introduced on the Cisco 2600 series and Cisco 3600 series.
12.0(7)T	This command was integrated into 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 implemented on the Cisco 1700 series, Cisco 2600XM, Cisco 3700 series, Cisco 7200 series, Cisco 7500 series, Cisco ICS 7750, Cisco MC3810, and Cisco VG200.

Usage Guidelines

Voice-card configuration mode is used for commands that configure the use of digital signal processor (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 and Cisco 3600 series routers and on high-performance compression modules on the Cisco MC3810. DSP resources are also found on some advanced integration modules (AIM-VOICE-30 and AIM-ATM-VOICE-30) on Cisco 2600 series and Cisco 3660 routers.

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

- On Cisco 2600 series and Cisco 3600 series routers, the *slot* argument corresponds to the physical chassis slot of the network module that has DSP resources to be configured.
- On the Cisco MC3810, the *slot* argument is always 0, and changes that are made in voice-card mode apply to the entire Cisco MC3810. On the Cisco MC3810, the **voice-card** command is available only if the chassis is equipped with one or two HCMs.

DSP resource sharing is also configured in voice-card configuration mode. On Cisco 2600 series and Cisco 3660 routers under specific circumstances, 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 **dspint dspfarm** command reference for more information about DSP resource sharing.

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

Examples

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

```
Router(config)# voice-card 1
```

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

```
Router(config)# voice-card 0
```

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-port

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

Cisco 1750 and Cisco 1751

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

Cisco 2600, Cisco 3600 Series and Cisco 7200 Series

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

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

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

Cisco AS5300

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

Cisco AS5800

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

Cisco MC3810

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

Syntax Description

Cisco 1750 and Cisco 1751

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

Cisco 2600, Cisco 3600 Series and Cisco 7200 Series

slot-number	Number of the slot in the router in which the VIC is installed. Valid entries are from 0 to 3, depending on the slot in which it has been installed.
subunit-number	Subunit on the VIC in which the voice port is located. Valid entries are 0 or 1.
port	Voice port number. Valid entries are 0 and 1.
slot	The router location in which the voice port adapter is installed. Valid entries are from 0 to 3.
*port***:**	Indicates the voice interface card location. Valid entries are 0 and 3.
ds0-group-no	Indicates the defined DS-0 group number. Each defined DS-0 group number is represented on a separate voice port. This allows you to define individual DS-0s on the digital T1/E1 card.

Cisco AS5300:

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

Cisco AS5800:

shelf	Specifies the T1 or E1 controller on the T1 card, or the T1 controller on the T3 card. Valid entries for the *shelf* argument are 0 to 9999.
slot	Specifies the T1 or E1 controller on the T1 card, or the T1 controller on the T3 card. Valid entries for the *slot* argument are 0 to 11.
port	Specifies the voice port number. • T1 or E1 controller on the T1 card —Valid entries are 0 to 11. • T1 controller on the T3 card—Valid entries are 1 to 28
:*port*	Specifies the value for the *parent* argument. The valid entry is 0.
:D	Indicates the D channel associated with ISDN PRI.

Cisco MC3810

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

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

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

Usage Guidelines

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

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

Examples

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

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

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

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

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

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

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

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

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

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

Related Commands

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

Glossary

AGM—Access Gateway Module. The Catalyst 4000 AGM extends the converged network to the branch office in an integrated LAN/WAN/VOICE platform.

CLR—Cell Loss Ratio.

CCS—common channel signaling.

DLCI—data-link connection identifier.

DSP—digital signal processor.

DTMF—dual-tone multifrequency. Tones generated when a button is pressed on a telephone; primarily used in the United States and Canada.

E&M—recEive and transMit (or ear and mouth).

EC, **ECAN**—echo canceller. A device placed in the four-wire portion of the circuit used for reducing near-end echo present on the send path by subtracting an estimation of that echo from the near-end echo. Note that an EC can also be used in an all-digital network.

echo path capacity—The maximum echo path delay for which an echo canceller is designed to operate.

echo path delay—The delay between the "receive out port Rout" and the "send in port Sin" ports of the echo canceller.

ERL—echo return loss. The attenuation of the signal between the receive out port Rout and the send in port Sin ports of the echo canceller.

ERLE—echo return loss Enhancement. The amount of echo attenuation provided by the echo canceller.

LMS, **NLMS**—ITU-T G.168 (2000) least mean square, normalized least mean square. Methods used to estimate the echo path model.

LR—Loudness Rating.

MMoIP—Multimedia Mail over IP. Dial peer specific to Store and Forward Fax. The MMoIP dial peer is the vehicle you use to assign particular line characteristics (such as a destination telephone number) to the connection between the Cisco router or the access server and the SMTP mail server during on-ramp faxing.

NLP—nonlinear processor. A component of the echo canceller that provides additional ERLE.

NM-HDA—High Density Analog Voice Network Module

PVC—permanent virtual circuit or, in ATM terminology, permanent virtual connection. Virtual circuit that is permanently established. PVCs save bandwidth associated with circuit establishment and are torn down in situations in which certain virtual circuits must exist all the time.

RLR—Receive Loudness Rating.

RSVP—Resource Reservation Protocol. Protocol that supports the reservation of resources across an IP network. Applications that are running on IP end systems can use RSVP to indicate to other nodes the nature (bandwidth, jitter, maximum burst, and so on) of the packet streams that they want to receive. RSVP depends on IPv6. Also known as Resource Reservation Setup Protocol.

RTOS—real time operating system.

SLR—Segmentation Local Reference.

TELR—Talker Echo Loudness Rating.

UDP—User Datagram Protocol. Connectionless transport layer protocol in the TCP/IP protocol stack. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols. UDP is defined in RFC 768.

VIC—voice interface card. Connects the system either to the PSTN or to a PBX. Compare with WIC.

WIC—WAN interface card. Connects the system to the WAN link service provider.