



Voice Call Tuning

The Voice Call Tuning feature monitors the interface between Cisco IOS software and a system's digital signaling processors (DSPs) in real time and reports status on the following: packet flow, DSP state, echo-cancellation state, and jitter state. The feature also allows you to manipulate echo-cancellation and jitter-buffer parameters in real time.

The feature is part of the standards product and is supported under normal Cisco Technical Assistance Center (TAC) procedures. No new code is required. It is backward compatible with older versions of Cisco IOS software, Cisco VCWare, and Cisco DSPWare, and accommodates the Conexant family of products.

Feature Specifications for Voice Call Tuning

Feature History

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

Supported Platforms

Cisco 1700 series, Cisco 2400 series, Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco 7500 series, Cisco 7750, Cisco 827, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, Cisco AS5850, Cisco CVA122, Cisco MC3810, Cisco uBR92x, Cisco VG200.

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 will verify 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.

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Restrictions for Voice Call Tuning

If you wish to use this feature on a Cisco AS5300 platform, refer to the *Combined Version Release Notes for Cisco VCWare on Cisco AS5300 Universal Access Servers/Voice Gateways* for more information about which Cisco VCWare to use with this feature.

Information About Voice Call Tuning

To configure the Voice Call Tuning feature, you must understand the following concepts:

- [Packet-Flow Detection, page 3](#)
- [DSP State, page 3](#)
- [Echo-Cancellation State, page 3](#)
- [Jitter-Buffer Parameters, page 3](#)

Packet-Flow Detection

The analog signal from the telephone is digitized into pulse code modulation (PCM) signals by the voice coder-decoder (codec). The PCM samples are then passed to the compression algorithm which compresses the voice into a packet format for transmission across the WAN. On the far side of the cloud the exact same functions are performed in reverse order.

Depending on how the network is configured, the router or gateway can perform both the codec and compression functions or only one of them. For example, if an analog voice system is used, then the router or gateway performs the CODEC function and the compression function.

The Voice Call Tuning feature allows you to determine whether packets are flowing in each direction while a call is in progress.

DSP State

The Voice Call Tuning feature allows you to check whether the DSP servicing an active call is functioning properly. Explicit configuration of a periodic ping message with both the period and timeout parameters configurable catches DSPs that are in a hung state as soon as possible. A failure can then signal this event to the console log or as a triggering event to a logging functionality. The DSP can also send periodic informational messages like jitter measures, echo canceller measures, and alarms to alert you if there is an impending problem.

Echo-Cancellation State

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

This feature gives you the ability to change echo canceller parameters while a call is in progress. Audible effects are immediately noticed, aiding in problem determination and resolution. You have the ability to change the echo canceller parameters (tail length and ERL threshold) in the field.

Jitter-Buffer Parameters

Jitter is a variation in the delay of received packets. At the sending side, packets are sent in a continuous stream with the packets being spaced evenly apart. Due to network congestion, improper queuing, or configuration errors, this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant.

This feature gives you the ability to change jitter-buffer parameters while a call is in progress. Audible effects are immediately noticed, aiding in problem determination and resolution. You have the ability to change the jitter buffer parameters (delay, size, fixed state or adaptive state) in the field.

How to Verify Voice Call Tuning Functionality

There are no configuration tasks for this feature. However, you can verify that the Voice Call Tuning feature is operating on your system by performing the following tasks:

-
- Step 1** Use the **show vfc version** command to show the version of the software that resides on your voice feature card (VFC). This command shows information in the output of the **show vfc slot version veware** and **show vfc slot version dspware** commands that indicates whether the Cisco VCWare or DSPWare is compatible with the Cisco IOS image.
- Step 2** Use the **show voice call** command to show the real-time call status for voice ports, including packet flow indication, DSP state, echo canceller state, and jitter state.
- Step 3** Use the **test call id** command to manipulate echo canceller and jitter-buffer parameters in real time. 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, which allows you to configure the router as a whole. Messages are visible in the command output when either an extended-only or a standard-only echo cancellation is requested:

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

Configuration Examples for Voice Call Tuning

There are no specific configuration examples for the Voice Call Tuning feature, but you can verify that the Voice Call Tuning feature is operating on your system by performing the following tasks:

- Use the **show vfc version** command to show the version of the software that resides on your voice feature card (VFC). This command was modified to include new information in the output of the **show vfc slot version veware** and **show vfc slot version dspware** commands for the Voice Call Tuning feature. Messages are output if the Cisco VCWare or DSPWare is not compatible with the Cisco IOS image. The new information is advisory only, so there is no action taken if the software is compatible or incompatible.

Cisco IOS Release 12.2(13)T adds new information to the output of the **show vfc slot version veware** and **show vfc slot version dspware** commands. Messages are output if the Cisco VCWare or DSPWare is not compatible with the Cisco IOS image. The new information is advisory only, so there is no action taken if the software is compatible or incompatible.

If the versions detected fall within the defined criteria and are compatible, nothing is output at bootup time. A confirmation line is output when the **show vfc version veware** and **show vfc version dspware** commands are used:

```
Router# show vfc 1 version veware

Voice Feature Card in Slot 1:
VCWare Version      : 7.35
ROM Monitor Version: 1.3
  DSPWare Version   : 3.4.46L
  Technology        : C549
VCWare/DSPWare version compatibility OK
```

- Use the **show voice call** command to show the real-time call status for voice ports, including packet flow indication, DSP state, echo canceller state, and jitter state. This command was modified with the **status**, **call-id**, and **sample sample-period** command options for the Voice Call Tuning feature. This command is available on all voice platforms.

Using the **call-id** argument is a generic means to identify active calls. If the **call-id** argument is omitted, the enquiry will show all active voice calls. In the following example, 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
```

- Use the **test call id** command to manipulate the echo canceller and jitter buffer parameters in real time. 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 example shows query output from a Cisco AS5350 universal gateway using the NextPort version of the standard echo canceller:

```
Router# test call id 99 ?

echo-canceller  test echo canceller on an active voice call
playout-delay   test playout-delay parameters
```

Additional References

For additional information related to the Voice Call Tuning feature, see the following sections:

- [Related Documents, page 6](#)
- [Standards, page 6](#)
- [MIBs, page 6](#)
- [RFCs, page 7](#)
- [Technical Assistance, page 7](#)

Related Documents

Related Topic	Document Title
How to configure your Cisco router or access server to support voice, video, and fax applications.	<i>Cisco IOS Voice, Video, and Fax Configuration Guide</i> , Release 12.2
How to use Cisco IOS commands to support voice, video, and fax applications.	<i>Cisco IOS Voice, Video, and Fax Command Reference</i> , Release 12.2 T
New features and changes to the voice feature card software (VCWare) and digital signal processor (DSP) firmware (DSPWare) on Cisco AS5300 universal voice gateways	<i>Combined Version Release Notes for Cisco VCWare on Cisco AS5300 Universal Access Servers/Voice Gateways</i> .
Voice ports on routers and access servers and how they emulate physical telephony switch connections so that voice calls and their associated signaling can be transferred intact between a packet network and a circuit-switched network or device	<i>Configuring Voice Ports</i> , Cisco IOS Release 12.2
Third-party extended echo canceller (EC) used in Cisco gateways with Cisco IOS software	<i>Extended ITU-T G.168 Echo Cancellation</i>
Enhancements to the playout-delay command, which configures the jitter buffer to reduce delay variation on a VoIP network	<i>Playout Delay Enhancements for Voice over IP</i>
New features in Cisco IOS Release 12.2 and 12.2-based releases	Release Notes index, Cisco IOS Release 12.2

Standards

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

MIBs

MIBs	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature.	To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB website on Cisco.com at the following URL: http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:

<http://tools.cisco.com/ITDIT/MIBS/servlet/index>

If Cisco MIB Locator does not support the MIB information that you need, you can also obtain a list of supported MIBs and download MIBs from the Cisco MIBs page at the following URL:

<http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml>

To access Cisco MIB Locator, 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 will verify 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>

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.	—

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

- [show vfc version](#)
- [show voice call](#)
- [test call id](#)

show vfc version

To show the version of the software that resides on a voice feature card (VFC), use the **show vfc version** command in privileged or user EXEC mode.

```
show vfc slot version {dspware | vcware}
```

Syntax Description	slot	Sot in which the VFC is installed. Valid values range from 0 to 2.
	dspware	Specifies the DSPWare version.
	vcware	Specifies the Cisco VCWare version.

Command Modes	Privileged or user EXEC
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Command History	Release	Modification
	11.3 NA	This command was introduced on the Cisco AS5300 universal access server.
	12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T with changes to the command output.

Usage Guidelines	Messages are output if the Cisco VCWare or DSPWare is not compatible with the Cisco IOS image. The new information is advisory only. You need take no action if the software is incompatible.
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Examples	If the versions detected fall within the defined criteria and are compatible, nothing is output at bootup time. A confirmation line is output when the show vfc version vcware and show vfc version dspware commands are used:
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```
Router# show vfc 1 version vcware

Voice Feature Card in Slot 1:
VCWare Version      : 7.35
ROM Monitor Version: 1.3
  DSPWare Version   : 3.4.46L
  Technology        : C549
VCWare/DSPWare version compatibility OK
```

```
Router# show vfc 1 version dspware

DSPWare version in VFC slot 1 is 3.4.46L
VCWare/DSPWare version compatibility OK
```

If the found versions are out of range or otherwise mismatched, a representative message is output when you boot up the router or is appended to the output of the commands. Other than the output of these messages, the version check has no other effect, and the software functions normally. The following example shows a found version that is out of range or mismatched at bootup:

```
...
Firmware version mismatch for bundle AS5300 VCWare
- version found (6.04) is lower than minimum required (7.35)
```

```
Firmware version mismatch for bundle AS5300 C549
- version found (3.3.10L) is lower than minimum required (3.4.46L)
```

If you were to enter an explicit request and the software was incompatible, the following appears:

```
Router# show vfc 1 version veware
```

```
Voice Feature Card in Slot 1:
VCWare Version      : 6.04
ROM Monitor Version: 1.3
  DSPWare Version   : 3.3.10L
  Technology        : C549
Firmware version mismatch for bundle AS5300 VCWare
- version found (6.04) is lower than minimum required (7.14)
Firmware version mismatch for bundle AS5300 C549
- version found (3.3.10L) is lower than minimum required (3.4.26L)
```

```
Router# show vfc 1 version dspware
```

```
DSPWare version in VFC slot 1 is 3.3.10L
Firmware version mismatch for bundle AS5300 VCWare
- version found (6.04) is lower than minimum required (7.14)
Firmware version mismatch for bundle AS5300 C549
- version found (3.3.10L) is lower than minimum required (3.4.26L)
```

[Table 1](#) Describes significant fields shown in the command output for the **show vfc version veware** command with compatible firmware.

Table 1 *show vfc version veware Field Descriptions*

Field	Description
VCWare Version	Cisco VCWare version
ROM Monitor version	ROM monitor version
DSPWare Version	DSPWare version
Technology	Technology available to support either medium-complexity codecs or high-complexity codecs
VCWare/DSPWare version compatibility	Cisco VCWare/DSPWare version compatibility is either OK or shows a mismatch. Note This option is available only with Cisco IOS Release 12.2(10) mainline and higher or Cisco IOS Release 12.2(11)T and higher.

Table 2 shows output field descriptions for the **show vfc version dspware** command with compatible firmware.

Table 2 *show vfc version dspware Field Descriptions*

Field	Description
DSPWare Version	DSPWare version
VCWare/DSPWare version compatibility	Cisco VCWare/DSPWare version compatibility is either OK or shows a mismatch. Note This option is available only with Cisco IOS Release 12.2(10) mainline and higher or 12.2(11)T and higher.

■ show vfc version

Related Commands

Command	Description
show vfc cap-list	Displays the current list of files on the capability list for this VFC.
show vfc default-file	Displays the default files included in the default file list for this VFC.
show vfc directory	Displays the list of all files residing on this VFC.

show voice call

To show the real-time call status for voice ports, including packet flow indication, DSP state, echo canceller state, and jitter state, use the **show voice call** command in either user EXEC or privileged EXEC mode.

Cisco 1700 Series and Cisco 7750 with Analog Voice Ports

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

Cisco 2600, Cisco 3600, and Cisco 3700 Series with Analog Voice Ports

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

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 | summary call-id | status] [sample sample-period]
```

Cisco 7200 and Cisco 7500 Series with Digital Voice Ports

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

Cisco 827 with Analog Voice Ports

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

Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850 with Analog Voice Ports

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

Cisco CVA122 with Analog Voice Ports

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

Cisco MC3810 with Analog Voice Ports

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

Cisco MC3810 with Digital Voice Ports

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

Cisco uBR92x with Analog Voice Ports

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

Cisco VG200 with Analog Voice Ports

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

Cisco VG200 with Digital Voice Ports

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

For the Cisco 1700 Series and Cisco 7750 with Analog Voice Ports:

<i>slot/subunit/port</i>	(Optional) Displays information for the analog voice port you specify with the <i>slot/subunit/port</i> designation. <ul style="list-style-type: none"> • <i>slot</i> specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • <i>subunit</i> specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) • <i>port</i> specifies an analog voice port number. Valid entries are 0 and 1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For Cisco 2600, Cisco 3600, and Cisco 3700 Series with Analog Voice Ports:

<i>slot/subunit/port</i>	(Optional) Displays information for the analog voice port you specify with the <i>slot/subunit/port</i> designation. <ul style="list-style-type: none"> • <i>slot</i> specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • <i>subunit</i> specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) • <i>port</i> specifies an analog voice port number. Valid entries are 0 and 1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For Cisco 2600, Cisco 3600, and Cisco Series with Digital Voice Ports:

<i>slot/port:ds0-group</i>	(Optional) Displays information for the digital voice port you specify with the <i>slot/port:ds0-group</i> designation. <ul style="list-style-type: none"> • <i>slot</i> 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. • <i>port</i> 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.) • <i>ds0-group</i> specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.

status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco 7200 Series with Digital Voice Ports:

<i>slot/port:ds0-group</i>	(Optional) Displays information for the digital voice port you specify with the <i>slot/port:ds0-group</i> designation. <ul style="list-style-type: none"> • <i>slot</i> 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. • <i>port</i> 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.) • <i>ds0-group</i> specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco AS5300 Series with Digital Voice Ports



Note

The Voice Call Tuning feature is not supported on the Cisco AS5300.:

<i>slot/port:ds0-group</i>	(Optional) Displays information for the digital voice port you specify with the <i>slot/port:ds0-group</i> designation. <ul style="list-style-type: none"> • <i>slot</i> 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. • <i>port</i> 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.) • <i>ds0-group</i> specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco AS5350 Series with Digital Voice Ports:

<i>slot/port:ds0-group</i>	(Optional) Displays information for the digital voice port you specify with the <i>slot/port:ds0-group</i> designation. <ul style="list-style-type: none"> • <i>slot</i> 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. • <i>port</i> 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.) • <i>ds0-group</i> specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco AS5400 Series with Digital Voice Ports:

<i>slot/port:ds0-group</i>	(Optional) Displays information for the digital voice port you specify with the <i>slot/port:ds0-group</i> designation. <ul style="list-style-type: none"> • <i>slot</i> 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. • <i>port</i> 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.) • <i>ds0-group</i> specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco AS5800 Series with Digital Voice Ports:

<i>slot/port:ds0-group</i>	(Optional) Displays information for the digital voice port you specify with the <i>slot/port:ds0-group</i> designation. <ul style="list-style-type: none"> • <i>slot</i> 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. • <i>port</i> 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.) • <i>ds0-group</i> specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.

status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco CVA122 with Digital Voice Ports:

<i>slot/port:ds0-group</i>	(Optional) Displays information for the digital voice port you specify with the <i>slot/port:ds0-group</i> designation. <ul style="list-style-type: none"> • <i>slot</i> 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. • <i>port</i> 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.) • <i>ds0-group</i> specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco MC3810 with Analog Voice Ports:

<i>slot/port</i>	(Optional) Displays information for the analog voice port you specify with the <i>slot/port</i> designation. <ul style="list-style-type: none"> • <i>slot</i> is the physical slot in which the analog voice module (AVM) is installed. The <i>slot</i> is always 1 for analog voice ports in the Cisco MC3810. • <i>port</i> specifies an analog voice port number. Valid entries are from 1 to 6.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco MC3810 with Digital Voice Ports:

<i>slot:ds0-group</i>	(Optional) Displays information for the digital voice port you specify with the <i>slot:ds0-group</i> designation. <ul style="list-style-type: none"> • <i>slot</i> specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). • <i>ds0-group</i> specifies a T1 or E1 logical voice port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.

status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco uBR92x with Analog Voice Ports:

<i>slot/subunit/port</i>	(Optional) Displays information for the analog voice port you specify with the <i>slot/subunit/port</i> designation. <ul style="list-style-type: none"> • <i>slot</i> specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • <i>subunit</i> specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) • <i>port</i> specifies an analog voice port number. Valid entries are 0 and 1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

For the Cisco VG200 with Analog Voice Ports:

<i>slot/subunit/port</i>	(Optional) Displays information for the analog voice port you specify with the <i>slot/subunit/port</i> designation. <ul style="list-style-type: none"> • <i>slot</i> specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • <i>subunit</i> specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) • <i>port</i> specifies an analog voice port number. Valid entries are 0 and 1.
summary	(Optional) Displays a summary of all voice ports.
<i>call-id</i>	(Optional) Represents a call ID.
status	(Optional) Displays status information of all voice ports.
sample <i>sample-period</i>	(Optional) Show status over this sampling interval. The <i>sample-period</i> is the amount of time in seconds. Range is from 1 to 30. Default is 10.

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, and Cisco 3600 series.
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 , <i>call-id</i> , and sample sample-period command options.

Usage Guidelines

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 for example, 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.



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.

CODEC and VAD are not displayed by the **show voice call** command because this information is in the summary display.

The **show voice call** command provides the status at these 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.

The **sample sample-period** is the number of seconds over which the status is to be determined, ranging from 1 to 30 seconds. If **sample sample-period** is omitted, a default of 10 seconds is used. The results of the run-time statistic queries are then analyzed and presented in a one-line summary format.

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
```

When using the **test call id** command, you must specify a call ID. The call ID is obtained by using the **show voice call status** command. The following is an example of how to obtain the call ID parameter that will be the value used in the *call-id* argument. The first parameter displayed in the output shows the call ID.

**Note**

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

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

Keyword	Description
append	<p>Appends redirected output to URL (URLs supporting append operation only)</p> <p>Subcommands for the append keyword include the following:</p> <ul style="list-style-type: none"> • 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	<p>Begin with the line that matches</p> <p>Subcommands for the append keyword include the following:</p> <ul style="list-style-type: none"> • LINE—Regular expression
exclude	<p>Exclude lines that match</p> <p>Subcommands for the append keyword include the following:</p> <p>LINE—Regular expression</p>
include	<p>Include lines that match</p> <p>Subcommands for the append keyword include the following:</p> <p>LINE—Regular expression</p>

redirect	<p>Redirect output to URL</p> <p>Subcommands for the append keyword include the following:</p> <ul style="list-style-type: none"> • 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
<hr/>	
tee	<p>Copy output to URL</p> <p>Subcommands for the append keyword include the following:</p> <ul style="list-style-type: none"> • 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

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

Examples

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

```

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

```

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

```

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 is a sample display from the **show voice call** command for analog voice ports on a Cisco MC3810:

```

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 a sample display of the **show voice call status** command on the 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 is a generic means to identify active calls. If the *call-id* is omitted, the enquiry shows all active voice calls. In 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
```

The following example shows echo return loss (ERL) reflector information where 3 is the *call-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
```

In this example, the value under the ERL column is a measure of the echo return loss (in dB) as reported by the DSP. The values under the Jitter column are the current values 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 using the command above. The output shows a new value under ERL/Reflctr. This is a millisecond measure of the time difference 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 example shows output using 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
```

```
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
```

The following 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
```

Table 3 describes significant fields shown for the **show voice call status** command.

Table 3 *show voice call status Field Descriptions*

Field	Description
CallID	The hexadecimal number under the CallID column is used for further enquiry. It is the monotonically increasing number that call control maintains for each call leg (ccCallID_t).
CID	CID is the conglomerate value derived from the GUID that appears in the show call active brief command.
ccVdb	The ccVdb pointer is the value that is displayed in many other debugs to identify these call legs.
ERL	Measure of the echo return loss (in dB) as reported by the DSP.
ERL/Reflctr	Millisecond measure of the time difference between the original signal and the loudest echo (peak reflector) as detected by the echo canceller.
Port	Refer to the voice port.
DSP/Ch	DSP and channel allocated to this call leg. The format of these values is platform dependent (for example, 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
Codec	Codec
Dial-peers	Dial-peer

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 manipulate echo-canceller and jitter-buffer parameters in real time, use the **test call id** command in privileged EXEC mode.

```
test call id call-id [echo-canceller [coverage range] [erl worst-case [0 | 3 | 6]] [h-register [clear
| freeze | thaw]]] | [playout-delay [fixed fixed-delay]] | [adaptive nominal-delay min-delay
max-delay]]
```

Syntax Description	
<i>call-id</i>	The hexadecimal ID of an active voice call. Values can be 0-FFFFFFFF.
echo-canceller	(Optional) Tests the echo canceller on an active voice call.
coverage <i>range</i>	(Optional) Tests echo canceller coverage in milliseconds. Valid values are 0, 8, 16, 24, 32, 48, 64, and 128. Specific default values depend on your echo canceller firmware: <ul style="list-style-type: none"> Standard echo canceller (Cisco proprietary G.165 EC)—8 Extended echo canceller—64 NextPort firmware—8 See the Usage Guidelines for more information about default values.
erl worst-case [0 3 6]	(Optional) Worst-case echo return loss (ERL), in decibels. Valid values are 0, 3, or 6. Default is 6. Note echo-canceller erl worst-case is 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]	clear —Clears a call in the extended echo canceller H-register freeze —Freezes a call in the extended echo canceller H-register thaw —Thaws a call in the extended echo canceller H-register
playout-delay	(Optional) Resets the playout buffering on the associated digital signal processors (DSPs) to the requested values. If fixed <i>fixed-delay</i> is selected, the first parameter only is required and used. If all three parameters are used, they are accepted, but the last two are ignored. If adaptive <i>nominal-delay min-delay max-delay</i> is selected, all three values are required and used.

fixed <i>fixed-delay</i>	(Optional) Tests the fixed playout-delay mode. Jitter buffer size does not adjust during a call; a constant playout delay is added. The <i>fixed-delay</i> argument is nominal delay in ms. Range is from 0 to 1500.
adaptive <i>nominal-delay min-delay max-delay</i>	(Optional) 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, <i>min-delay</i> , <i>nominal-delay</i> , and <i>max-delay</i> 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. Minimum delay range is from 10 to 80. Nominal delay range is from 0 to 1500. Maximum delay range is from 40 to 1700. 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

Obtain the *call-id* argument value by using the **show voice call status** command. The following example shows how to obtain the call ID. The first parameter displayed in the output shows the call ID.

**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.

```
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 running the extended echo canceller. Table 4 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. 0 indicates a disabled state.

Table 4 Echo Canceller Types and Canceller Coverage Ranges

Platform	Echo Canceller Type	Echo Canceller Coverage Range
Cisco 1700 series	Standard	Standard—0, 8, 16, 24, 32
Cisco 2400 series	Standard	Standard—0, 8, 16, 24, 32
Cisco 2600 series	Standard, extended	Standard—0, 8, 16, 24, 32 Extended—0, 24, 32, 48, 64

Table 4 *Echo Canceller Types and Canceller Coverage Ranges (continued)*

Platform	Echo Canceller Type	Echo Canceller Coverage Range
Cisco 3600 series	Standard, extended	Standard—0, 8, 16, 24, 32 Extended—0, 24, 32, 48, 64
Cisco 7200	Standard, extended	Standard—0, 8, 16, 24, 32 Extended—0, 24, 32, 48, 64
Cisco 7750	Standard	Standard—0, 8, 16, 24, 32
Cisco 827	Standard	Standard—0, 8, 16, 24, 32
Cisco AS5300	Standard, extended	Standard—0, 8, 16, 24, 32 Extended—0, 24, 32, 48, 64
Cisco AS5350	NextPort	NextPort—0, 8, 16, 24, 32, 64, 128
Cisco AS5400	NextPort	NextPort—0, 8, 16, 24, 32, 64, 128
Cisco AS5800	Standard	Standard—0, 8, 16, 24, 32
Cisco AS5850	NextPort	NextPort—0, 8, 16, 24, 32, 64, 128
Cisco CVA122	Standard	Standard—0, 8, 16, 24, 32
Cisco MC3810	Standard, extended	Standard—0, 8, 16, 24, 32 Extended—0, 24, 32, 48, 64
Cisco uBR92x	Standard	Standard—0, 8, 16, 24, 32

Specified parameters are 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, allows you to configure the router as a whole. The following two new output might appear 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
```

You can display 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, the 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. jjj

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 shows how to set the fixed delay to 0, which is the minimum value allowed:

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

The following example shows how to set the minimum delay, nominal delay, and maximum delay. In this example, the maximum value allowed for each parameter is implemented:

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

The following example shows how to test 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 shows how to test 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:

```
Router# 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.

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

worst-case test extended echo canceller worst-case erl

Router# test call id 99 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 ERL
```

The following is example 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
```

Related Commands

Command	Description
show call threshold	Displays enabled triggers, current values for configured triggers, and number of API calls that were made to global and interface resources.
show call treatment	Displays the call treatment configuration and the statistics for handling the calls based upon resource availability.
show voice call	Shows the real-time call status for voice ports.
test call fallback probe	Tests current network conditions to a particular IP address and displays the ICPIFSAA values.
test call threshold	Tests how the core APIs-behave on the basis of the resource configuration.

Glossary

API—application programming interface.

CLI—command-line interface.

DSP—digital signal processor.

DSPWare—The total load of code that runs in a DSP for voice processing. The code is embedded in the Cisco IOS image for a platform.

HPI—host port interface, used with the TI C54x class of DSP.

PCM—pulse code modulation.

RTP—Real-Time Transport Protocol.

TDM—time-division multiplexing.

TI C54x—Texas Instruments C54x family of DSPs, including the C542, C549, C5409, C5410.

VCWare—Code image that runs on each VFC in a Cisco AS5300 system. Embeds the DSPWare that the Cisco AS5300 uses.

VFC—voice feature card.

