Voice Port Enhancements in Cisco 2600 and 3600 Series Routers and MC3810 Series Concentrators

This document describes enhancements introduced in Cisco IOS Release 12.0(7)XK that extend the cross-platform commonality of voice port configuration procedures on the Cisco 2600 and 3600 series routers and MC3810 series concentrators.

This document includes the following sections:

Feature Summary

The Cisco 2600 series and 3600 series routers and Cisco MC3810 series multiservice access concentrators support data, voice, and video transport to varying degrees. Numerous voice port commands and features that were previously limited to one or two of these platforms have been extended to additional platforms, and differences in configuration commands have been reduced or eliminated.

Benefits

These enhancements provide the following improvements to the platforms involved:

- Increase the voice capabilities of the platforms gaining new features
- Increase the level of interoperability between the platforms
- Simplify the configuration procedures

Restrictions

None

Related Documents

- *Voice over IP on the Cisco MC3810*, Cisco IOS Release 12.0(7)XK Online Document
- *Voice over ATM on the Cisco 2600 and 3600 Routers*, Cisco IOS Release 12.0(7)XK Online Document

Supported Platforms

- Cisco 2600 series
- Cisco 3600 series
- Cisco MC3810 series

Supported Standards, MIBs, and RFCs

TIA-EIA 464-B—Requirements for Private Branch Exchange (PBX) Switching Equipment

Prerequisites

The voice enhancements described in this document require the use of Cisco IOS Release 12.0(7)XK or later.

Configuration Tasks

Most voice-port configuration commands for the Cisco 2600, 3600, and MC3810 series platforms have been made usable on all three platforms. Differences in usage are noted for individual commands in the command reference section.

This document describes new and changed procedures applicable to voice ports for Voice over IP (VoIP), Voice over Frame Relay (VoFR), and Voice over ATM (VoATM) on Cisco 2600 and 3600 series routers and Cisco MC3810 series concentrators. Commands apply to both analog and digital voice ports unless otherwise indicated.

- Configuring DSP Bypass Options
- Configuring Permanent Connection Options
- Configuring Ring Cadence
- Configuring Auto-Cut-Through Options
- Configuring E&M Signaling Bit Functioning
- Configuring Gain Offset
- Manipulating Signaling Bits
- Configuring Disconnect Acknowledgment
- Configuring Playout Delay
- Configuring Voice-Port Timing Characteristics

- Configuring Battery Reversal
- Configuring FXS Port Idle Voltage
- Using Voice-Related Show Commands

Configuring DSP Bypass Options

This section describes how to control whether local calls bypass the DSP or go through the DSP. Local calls normally bypass the DSP to minimize use of system resources (the default). Use this procedure to direct local calls through the DSP or to restore the default (DSP bypass).

To enable the input gain and output attenuation functions on a router or concentrator, you must disable voice local bypass.

Disabling Voice Local Bypass

(Cisco MC3810 only) To pass local calls through the DSPs, enter the following commands beginning in global configuration mode. If local calls are processed through the DSPs, the DSPs provide ringback tone to the voice ports.

Step	Command	Purpose
1	`router(config)#` **`no voice local-bypass`**	Configures local calls to be processed through the DSPs.

Enabling Voice Local Bypass

(Cisco MC3810 only) To restore the default configuration, in which local calls bypass the DSPs, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	`router#` **`configure terminal`**	Enter global configuration mode.
2	`router(config)#` **`voice local-bypass`**	Configures local calls to bypass the DSPs.

Configuring Permanent Connection Options

This section describes how to configure a voice-port connection mode and destination telephone number for permanent connections. This feature was unified across the Cisco MC3810, 2600, and 3600 platforms in Cisco IOS Release 12.0(7)XK.

To configure a connection mode and destination telephone number for a permanent connection through a voice port, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#` **`connection`** {**`plar`** | **`tie-line`** | **`plar-opx`**} *`digits`* {**`trunk`** *`digits`* [**`answer-mode`**]}	Specify the voice-port connection type and the destination telephone number. • **plar** for private line automatic ringdown • **tie-line** for a tie-line connection to a PBX • **plar-opx** for PLAR off-premises extension (the local voice port provides a local response before the remote voice port receives an answer) • **trunk** for a straight tie-line connection to a PBX • **answer-mode** if a trunk connection is specified and the router should not attempt to initiate a trunk connection, but should wait for an incoming call before establishing the trunk • *digits* specifies the destination telephone number.
3	`router(config-voiceport)#` **`voice confirmation-tone`**	If **connection plar** or **connection plar-opx** is configured, enable the two-beep confirmation tone that a caller hears when picking up the handset.

Configuring Ring Cadence

This section describes how to specify on and off times for ringing pulses on an FXS voice port. The ability to specify ring cadence is a new feature on the Cisco 2600 and 3600 platforms, and the syntax for configuring the ring cadence is new in IOS Release 12.0(7)XK.

To configure ring cadence, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.

Step	Command	Purpose
2	router(config-voiceport)# **ring cadence** {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] [**define** *pulse-interval*]}	(FXS only) Specify the on and off times for the ringing pulses. See the command reference section for details on the ring cadence options.

Configuring Auto-Cut-Through Options

This section describes how to disable or enable the auto-cut-through feature on E&M voice ports. When enabled, this feature makes call completion possible when a PBX does not provide an M-lead response. This feature is enabled by default on E&M voice ports. This is a new feature on the Cisco 2600 and 3600 platforms in Cisco IOS Release 12.0(7)XK.

Disabling Auto-Cut-Through

To disable auto-cut-through on an E&M voice port, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: router(config)# **voice-port** *slot/subunit/port* For Cisco 2600 and 3600 series digital voice ports: router(config)# **voice-port** *slot/port:ds0-group* For Cisco MC3810 series analog voice ports: router(config)# **voice-port** *slot/port* For Cisco MC3810 series digital voice ports: router(config)# **voice-port** *slot:ds0-group*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	router(config-voiceport)# **no auto-cut-through**	Disable the auto-cut-through feature.

Enabling Auto-Cut-Through

To enable auto-cut-through on an E&M voice port, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: router(config)# **voice-port** *slot/subunit/port* For Cisco 2600 and 3600 series digital voice ports: router(config)# **voice-port** *slot/port:ds0-group* For Cisco MC3810 series analog voice ports: router(config)# **voice-port** *slot/port* For Cisco MC3810 series digital voice ports: router(config)# **voice-port** *slot:ds0-group*	Identify the voice port you want to configure and enter voice-port configuration mode.

Step	Command	Purpose
2	router(config-voiceport)# **auto-cut-through**	Enable the auto-cut-through feature if it was previously disabled.

Configuring E&M Signaling Bit Functioning

This section describes how to modify the functioning of transmit and receive signaling bits for E&M and E&M MELCAS voice signaling. These are new features on the Cisco 2600 and 3600 series routers in IOS Release 12.0(7)XK.

Enter the following commands beginning in global configuration mode, to:

- Define transmit and receive signaling bits for E&M and E&M MELCAS voice signaling, if patterns different from the preset defaults are required. See the command reference for the default signaling patterns as defined in ANSI and CEPT standards.
- Specify which receive bit an E&M or E&M MELCAS voice port monitors and which receive bits it ignores, if patterns different from the defaults are required.

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: router(config)# **voice-port** *slot/subunit/port* For Cisco 2600 and 3600 series digital voice ports: router(config)# **voice-port** *slot/port:ds0-group* For Cisco MC3810 series analog voice ports: router(config)# **voice-port** *slot/port* For Cisco MC3810 series digital voice ports: router(config)# **voice-port** *slot:ds0-group*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	router(config-voiceport)# **define** {**Tx-bits** | **Rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | | **1101** | **1110** | **1111**}	(For T1/E1 digital voice ports only.) Define specific transmit and/or receive signaling bits to match the bit patterns required by a connected device.
3	router(config-voiceport)# **ignore** {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}	(For T1/E1 digital voice ports only.) Configure the voice port to ignore one receive bit.
4	router(config-voiceport)# **ignore** {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}	(For T1/E1 digital voice ports only.) Configure the voice port to ignore a second receive bit.
5	router(config-voiceport)# **ignore** {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}	(For T1/E1 digital voice ports only.) Configure the voice port to ignore a third receive bit.
6	router(config-voiceport)# **no ignore** {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}	(For T1/E1 digital voice ports only.) Configure the voice port to monitor one receive bit.

Configuring Gain Offset

This section describes how to specify a gain offset for the analog voice signal between an FXS or FXO analog voice port and the digital signal processor (DSP). This feature makes it possible to compensate for different signal levels from a PBX or DSP. The gain offset feature is available only on Cisco MC3810 series concentrators.

To configure the gain offset for an FXS or FXO voice port, enter the following commands, beginning in privileged EXEC mode:

Step	Command	Purpose
1	For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#` **`loss-plan`** {**`plan1`** | **`plan2`** | **`plan3`** | **`plan4`** | **`plan5`** | **`plan6`** | **`plan7`** | **`plan8`** | **`plan9`**}	Specify the loss plan for this voice port according to the signal level requirements for the DSP and the PBX. The default is **plan1**, which provides the following gain offset levels: • FXO—A-D gain = 0 dB, D-A gain = 0 dB • FXS—A-D gain = –3 dB, D-A gain = –3 dB
3	`router(config-voiceport)#` **`exit`**	Exit from voice-port configuration mode.

Manipulating Signaling Bits

This section describes how to force individual transmit and receive signaling bit states on any voice port type. This is a new feature on the Cisco 2600 and 3600 series routers in IOS Release 12.0(7)XK.

To force transmit and/or receive bit to an on, off, or inverted state, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#` **`condition`** {**`tx-a-bit`** | **`tx-b-bit`** | **`tx-c-bit`** | **`tx-d-bit`**} {**`rx-a-bit`** | **`rx-b-bit`** | **`rx-c-bit`** | **`rx-d-bit`**} {**`on`** | **`off`** | **`invert`**}	Configure the voice port to manipulate a transmit or receive bit pattern to match the bit pattern required by a connected device. Repeat the command for each transmit and/or receive bit to be modified. Be careful not to destroy the information content of the bit pattern. For example, forcing the A-bit on or off will prevent FXO interfaces from being able to generate an off-hook or on-hook state. **Note** The **show voice port** command reports at the protocol level, while the **show controller** command reports at the driver level. The driver is not notified of any bit manipulation using the **condition** command. As a result, the **show controller** command output will not account for the bit conditioning.

Configuring Disconnect Acknowledgment

This section describes how to configure an FXS or FXS MELCAS voice port to return an acknowledgment upon receipt of a disconnect signal. This is a new feature on the Cisco 2600 and 3600 series routers in IOS Release 12.0(7)XK.

To configure disconnect acknowledgment on an FXS voice port, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#` **`disconnect-ack`**	Configure the FXS voice port to return an acknowledgment upon receipt of a disconnect signal. The FXS port will remove line power if the equipment on the FXS loop-start trunk disconnects first.

Configuring Playout Delay

This section describes how to tune the playout buffer to accommodate packet jitter caused by switches in the WAN. This is a new feature on the Cisco 2600 and 3600 series routers in Cisco IOS Release 12.0(7)XK.

To change the maximum and/or nominal playout delay values on a voice port if the default values do not accommodate the jitter, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#` **`playout-delay maximum`** *`milliseconds`*	Configure the maximum playout delay time. The range is 40 to 320 milliseconds.
3	`router(config-voiceport)#` **`playout-delay nominal`** *`milliseconds`*	Configure the nominal playout delay time. The range is 40 to 240 milliseconds.

Configuring Voice-Port Timing Characteristics

This section describes how to change various timing characteristics on voice port. These are new features on the Cisco 2600 and 3600 series routers in IOS Release 12.0(7)XK.

Configuring Guard-Out Time on FXO Voice Ports

To change the guard-out duration of an FXO voice port, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#` **`timing guard-out`** *`milliseconds`*	Specify the duration in milliseconds of the guard-out period to prevent this port from seizing a remote FXS port before the remote port detects a disconnect signal. The range is 300 to 3000. The default is 2000.

Changing the Timing Percent Break of Dialing Pulses

To change the percentage of the break period for dialing pulses for a voice port, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#` **`timing percentbreak`** *`percent`*	Specify the percentage of the break period for the dialing pulses, if different from the default. The range is 20 to 80. The default is 50.

Changing the Ringing Timeout on a Voice Port

To change the length of time that a caller can continue ringing a telephone when there is no answer, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#` **`timeouts ringing`** {*`seconds`* | **`infinity`**}	Specify the duration that the voice port allows ringing to continue if a call is not answered, or enter **infinity** if you want ringing to continue until the caller goes on hook. If you specify *seconds*, the range is 5 to 60000. The default is 180.

Changing the Wait Release Delay on a Voice Port

To change the delay timeout before the system starts the process for releasing a voice port, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	`router(config-voiceport)#` **`timeouts wait-release`** {*`seconds`* | **`infinity`**}	Specify the duration that a voice port stays in the call-failure state while the Cisco router or concentrator sends a busy tone, reorder tone, or an out-of-service tone to the port, or enter **infinity** if you want voice port not to be released as long as the call-failure state remains. If you specify *seconds*, the range is 3 to 3600. The default is 30.

Configuring the VAD Silence Detection Time

To change the minimum silence detection time for voice activity detection (VAD), enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	`router(config)#` **`voice vad-time`** *`seconds`*	Specify the delay time in milliseconds for silence detection and suppression of voice packet transmission. The range is 250 to 65535. The default is 250.

Configuring Battery Reversal

This section describes how to change the battery-reversal functions for FXO and FXS voice ports. This is a new feature on the Cisco MC3810, 2600, and 3600 platforms in Cisco IOS Release 12.0(7)XK.

To configure an FXO voice port not to disconnect when it detects a second battery reversal, or to configure an FXS voice port not to reverse its battery when it connects a call, enter the following commands beginning in global configuration mode:

Step	Command	Purpose
1	For Cisco 2600 and 3600 series analog voice ports: `router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode. **Note** On Cisco 2600 and 3600 series routers, only analog voice ports in VIC-2FXO-M1 and VIC-2FXO-M2 voice interface cards are able to detect battery reversal. Analog voice ports in VIC-2FXO and VIC-2FXO-EU voice interface cards do not detect battery reversal.
2	`router(config-voiceport)#` **`no battery-reversal`**	FXO—Configure a loopstart voice port not to disconnect when it detects a second battery reversal. FXS—Configure the voice port not to reverse battery when it connects calls.

Configuring FXS Port Idle Voltage

This section describes how to set the talk-battery idle voltage on FXS analog voice ports in Cisco MC3810 series concentrators. This was a new feature on the Cisco MC3810 series in Cisco IOS Release 12.0(4)T.

To specify the idle voltage on an FXS analog voice port, complete the following steps beginning in global configuration mode:

Step	Command	Purpose
1	router(config)# **voice-port** *slot/port*	Identify the voice port you want to configure and enter voice-port configuration mode.
2	router(config-voiceport)# **idle-voltage** {**high** | **low**}	Set the idle voltage on the FXS voice port to be high (–48V) or low (–24V) when the voice port is idle.

Using Voice-Related Show Commands

This section describes how to display configuration, call-processing, and state-machine information about voice ports. These commands have enhanced functionality on the Cisco 2600 and 3600 series routers in IOS Release 12.0(7)XK.

Displaying Voice Port Information

To display voice-port related configuration information, enter the following commands beginning in user EXEC or privileged EXEC mode:

Command	Purpose
For Cisco 2600 and 3600 series with analog voice ports: router# **show voice port** [*slot/subunit/port* | **summary**] For Cisco 2600 and 3600 series with digital voice ports: router# **show voice port** [*slot/port:ds0-group.ds0* | **summary**] For Cisco MC3810 series with analog voice ports: router# **show voice port** [*slot/port* | **summary**] For Cisco MC3810 series with digital voice ports: router# **show voice port** [*slot:ds0-group.ds0* | **summary**]	To display voice port configuration information for a specific voice port, enter the applicable voice-port. To display a summary of the configurations for all voice ports on the router or concentrator, enter the **summary** keyword.

Displaying Voice Call Information

To display voice-call information, enter the following commands beginning in user EXEC or privileged EXEC mode:

Command	Purpose
For Cisco 2600 and 3600 series with analog voice ports: router# **show voice port** [*slot/subunit/port* | **summary**] For Cisco 2600 and 3600 series with digital voice ports: router# **show voice port** [*slot/port:ds0-group.ds0* | **summary**] For Cisco MC3810 series with analog voice ports: router# **show voice port** [*slot/port* | **summary**] For Cisco MC3810 series with digital voice ports: router# **show voice port** [*slot:ds0-group.ds0* | **summary**]	To display voice call information for a specific voice port, enter the applicable voice-port. To display a summary of the call information for all voice ports on the router or concentrator, enter the **summary** keyword.

Displaying Voice Channel DSP Information

To display voice-channel DSP configuration information, enter the following commands beginning in user EXEC or privileged EXEC mode:

Command	Purpose
`router#` **`show voice dsp`**	Displays voice-channel configuration information for all DSP channels.

Displaying the Active Voice Call Table

To display the contents of the active call table, which shows all of the calls currently connected through the router or concentrator, enter the following commands beginning in user EXEC or privileged EXEC mode:

Command	Purpose
`router#` **`show call active voice`**	Shows all of the calls currently connected through the router or concentrator.

Displaying the Voice Call History Table

To display the contents of the call history table, enter the following commands beginning in user EXEC or privileged EXEC mode:

Command	Purpose
`router#` **`show call history voice`** [**`last`** *`number`* | **`brief`**]	Displays a listing of all voice calls connected through this router or concentrator in descending time order. Display the last calls connected through this router if you enter the keyword **last**, and define the number of calls to be displayed with the argument *number*. Displays a shortened version of the call history table if you use the keyword **brief**.

Command Reference

This section documents new or modified commands. Modified commands are indicated by an asterisk (*). All other commands used on these platforms are documented in the Cisco IOS Release 12.0 command reference publications.

- **auto-cut-through***
- **battery-reversal**
- **codec (voice-port)***
- **condition***
- **connection***
- **define***
- **disconnect-ack***
- **idle-voltage**
- **ignore***
- **loss-plan**
- **playout delay***
- **ring cadence***
- **show call active voice***
- **show call history voice***
- **show voice call***
- **show voice dsp***
- **show voice port***
- **timeouts ringing**
- **timeouts wait-release***
- **timing guard-out***
- **timing percentbreak***
- **voice local-bypass**
- **voice vad-time**

auto-cut-through

To enable call completion when a PBX does not provide an M-lead response, use the **auto-cut-through** voice-port configuration command. Use the **no** form of this command to disable the auto-cut-through operation.

auto-cut-through
no auto-cut-through

Syntax Description

This command has no arguments or keywords.

Defaults

Auto-cut-through is enabled.

Command Mode

Voice-port configuration

Command History

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

Usage Guidelines

The **auto-cut-through** command applies to E&M voice ports only.

Examples

The following example enables call completion on a Cisco MC3810 when a PBX does not provide an M-lead response:

```
router(config)# voice-port 1/1
router(config-voiceport)# auto-cut-through
```

The following example enables call completion on a Cisco 2600 or 3600 when a PBX does not provide an M-lead response:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# auto-cut-through
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.

battery-reversal

To specify battery polarity reversal on an FXO or FXS port, use the **battery-reversal** voice-port configuration command. Use the **no** form of this command to disable battery reversal.

battery-reversal
no battery-reversal

Syntax Description

This command has no arguments or keywords.

Defaults

Battery reversal is enabled.

Command Mode

Voice-port configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

The **battery-reversal** command applies to FXO and FXS voice ports. On Cisco 2600 and 3600 series routers, only analog voice ports in VIC-2FXO-M1 and VIC-2FXO-M2 voice interface cards are able to detect battery reversal; analog voice ports in VIC-2FXO and VIC-2FXO-EU voice interface cards do not detect battery reversal. On digital voice ports, battery reversal is only supported on E1 MELCAS; it is not supported in T1 channel associated signaling (CAS) or E1 CAS.

FXS ports normally reverse battery upon call connection. If an FXS port is connected to an FXO port that does not support battery reversal detection, you can use the **no battery-reversal** command on the FXS port to prevent unexpected behavior.

FXO ports in loopstart mode normally disconnect calls when they detect a second battery reversal (back to normal). You can use the **no battery-reversal** command on FXO ports to disable this action.

The **battery-reversal** command restores voice ports to their default battery-reversal operation.

Examples

The following example disables battery reversal on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
   router(config-voiceport)# no battery-reversal
```

The following example disables battery reversal on voice port 1/0/0 on a Cisco 2600 or 3600 series router:

```
router(config)# voice-port 1/0/0
   router(config-voiceport)# no battery-reversal
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.

codec (voice-port)

The **codec** voice-port configuration command on the Cisco MC3810 is no longer supported beginning in this release. The command was first supported in Cisco IOS Release 11.3(1)MA. Configure the codec value using the **codec** dial-peer configuration command.

condition

To manipulate the signaling format bit-pattern for all voice signaling types, use the **condition** command. Use the **no** form of this command to turn off conditioning on the voice port.

condition {**tx-a-bit** | **tx-b-bit** | **tx-c-bit** | **tx-d-bit**} {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**} {**on** | **off** | **invert**}
no condition {**tx-a-bit** | **tx-b-bit** | **tx-c-bit** | **tx-d-bit**} {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**} {**on** | **off** | **invert**}

Syntax Description

tx-a-bit	Transmit A bit.
tx-b-bit	Transmit B bit.
tx-c-bit	Transmit C bit.
tx-d-bit	Transmit D bit.
rx-a-bit	Receive A bit.
rx-b-bit	Receive B bit.
rx-c-bit	Receive C bit.
rx-d-bit	Receive D bit.
on	Forces the bit state to be 1.
off	Forces the bit state to be 0.
invert	Inverts the bit state.

Defaults

The signaling format is not manipulated (for all transmit or receive A, B, C, and D bits).

Command Mode

Voice-port configuration

Command History

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

Usage Guidelines

Use the **condition** command to manipulate the sent or received bit patterns to match expected patterns on a connected device. Be careful not to destroy the information content of the bit pattern. For example, forcing the A-bit on or off will prevent FXO interfaces from being able to generate both an on-hook and off-hook state.

Examples

The following example manipulates the signaling format bit-pattern on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
router(config-voiceport)# condition tx-a-bit invert
router(config-voiceport)# condition rx-a-bit invert
```

The following example manipulates the signaling format bit-pattern on voice port 1/1/2 on a Cisco 2600 or 3600:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# condition tx-a-bit invert
router(config-voiceport)# condition rx-a-bit invert
```

Related Commands

Command	Description
define	Defines the transmit and receive bits for E&M and E&M MELCAS voice signaling.
ignore	Configures the E&M or E&M MELCAS voice port to ignore specific receive bits.

connection

To specify a connection mode for a voice port, use the **connection** voice-port configuration command. Use the **no** form of this command to disable the selected connection mode.

connection {**plar** | **tie-line** | **plar-opx**} *digits* | {**trunk** *digits* [**answer-mode**]}
no connection {**plar** | **tie-line** | **plar-opx**} *digits* | {**trunk** *digits* [**answer-mode**]}

Syntax Description

plar	Specifies a private line automatic ring down (PLAR) connection. PLAR is an autodialing mechanism that permanently associates a voice interface with a far-end voice interface, allowing call completion to a specific telephone number or PBX without dialing. When the calling telephone goes off hook a predefined network dial peer is automatically matched, which sets up a call to the destination telephone or PBX.
tie-line	Specifies a connection that emulates a temporary tie-line trunk to a private branch exchange (PBX). A tie-line connection is automatically set up for each call and torn down when the call ends.
plar-opx	Specifies a PLAR Off-Premises eXtension connection. Using this option, the local voice port provides a local response before the remote voice port receives an answer. On FXO interfaces, the voice port will not answer until the remote side answers.
trunk	Specifies a connection that emulates a permanent trunk connection to a private branch exchange (PBX). A trunk connection remains “nailed up” in the absence of any active calls.
digits	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.
answer-mode	(Optional; used only with the **trunk** keyword.) Specifies that the router should not attempt to initiate a trunk connection, but should wait for an incoming call before establishing the trunk.

Defaults

No connection mode is specified.

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA1	This command was first supported on the Cisco MC3810, and the **tie-line** keyword was first made available on the Cisco MC3810.
11.3(1)MA5 and 12.0(2)T	The **plar-opx** keyword was first made available on the Cisco MC3810 as the **plar-opx-ringrelay** keyword. The keyword was shortened in a subsequent release.
12.0(3)XG	The **trunk** keyword was made available on the Cisco MC3810. The **trunk answer-mode** option was added.
12.0(7)XK	This command was unified across the Cisco 2600, 3600, and MC3810 platforms.

Usage Guidelines

Use this command to specify a connection mode for a specific interface. For example, use the **connection plar** command to specify a PLAR interface. The string you configure for this command is used as the called number for all incoming calls over this connection. The destination peer is determined by the called number.

Use the **connection trunk** command to specify a permanent, “nailed up” tie-line connection to a PBX. You can use the **connection trunk** command for E&M-to-E&M trunks, FXO-to-FXS trunks, and FXS-to-FXS trunks. Signaling will be transported for E&M-to-E&M trunks and FXO-to-FXS trunks; signaling will not be transported for FXS-to-FXS trunks.

To configure one of the devices in the trunk connection to act as slave and only receive calls, use the **answer-mode** option with the **connection trunk** command when configuring that device.

Note When using the **connection trunk** command, you must perform a **shutdown/no shutdown** command sequence on the voice port.

Use the **connection tie-line** command when the dial plan requires that additional digits be added in front of any digits dialed by the PBX, and that the combined set of digits be used to route the call onto the network. The operation is similar to the **connection plar** command operation, but in this case the tie-line port waits to collect digits from the PBX. The tie-line digits are automatically stripped by a terminating port.

If the **connection** command is not configured, the standard session application outputs a dial tone when the interface goes off-hook until enough digits are collected to match a dial-peer and complete the call.

Examples

The following example selects PLAR as the connection mode on a Cisco 3600, with a destination telephone number of 555-9262:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# connection trunk 5559262
```

The following example selects tie-line as the connection mode on a Cisco MC3810, with a destination telephone number of 555-9262:

```
router(config)# voice-port 1/1
router(config-voiceport)# connection tie-line 5559262
```

The following example specifies a PLAR off-premises extension connection on a Cisco 3600, with a destination telephone number of 555-9262:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# connection plar-opx 5559262
```

The following example configures a Cisco 3600 series router for a trunk connection and specifies that it will establish the trunk only when it receives an incoming call:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# connection trunk 5559262 answer-mode
```

Related Commands

Command	Description
session-protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session-target	Configures a network-specific address for a dial peer.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.
destination-pattern	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.

define

To define the transmit and receive bits for E&M and E&M Mercury Exchange Limited (MELCAS) voice signaling, use the **define** voice-port configuration command. Use the **no** form of this command to restore the default value.

define {**Tx-bits** | **Rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}
no define {**Tx-bits** | **Rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}

Syntax Description

Tx-bits	Transmit signaling bits.
Rx-bits	Receive signaling bits.
seize	The bit pattern defines the seized state.
idle	The bit pattern defines the idle state.
0000 through **1111**	Specifies the bit pattern.

Defaults

The default is to use the preset signaling patterns as defined in ANSI and CEPT standards, as follows:

For E&M:

Tx-bits idle 0000 (0001 if on E1 trunk)
Tx-bits seize 1111
Rx-bits idle 0000
Rx-bits seize 1111

For E&M MELCAS:

Tx-bits idle 1101
Tx-bits seize 0101
Rx-bits idle 1101
Rx-bits seize 0101

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1) MA3	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

This command applies to E&M digital voice ports associated with T1/E1 controllers.

Use the **define** command to match the E&M bit patterns with the attached telephony device. Be careful not to define invalid configurations, such as all 0000 on E1, or identical seized and idle states. Use this command with the **ignore** command.

Examples

To configure a voice port on a Cisco 2600 or 3600 router sending traffic in North American E&M signaling format to convert the signaling to MELCAS format, enter the following commands:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# define rx-bits idle 1101
router(config-voiceport)# define rx-bits idle 0101
router(config-voiceport)# define tx-bits seize 1101
router(config-voiceport)# define tx-bits seize 0101
```

To configure a voice port on a Cisco MC3810 sending traffic in North American E&M signaling format to convert the signaling to MELCAS format, enter the following commands:

```
router(config)# voice-port 0/8
router(config-voiceport)# define rx-bits idle 1101
router(config-voiceport)# define rx-bits idle 0101
router(config-voiceport)# define tx-bits seize 1101
router(config-voiceport)# define tx-bits seize 0101
```

Related Commands

Command	Description
condition	Manipulate the signaling bit-pattern for all voice signaling types.
ignore	Configures an E&M or E&M MELCAS voice port to ignore specific receive bits.

disconnect-ack

To configure an FXS voice port to return an acknowledgment upon receipt of a disconnect signal, use the **disconnect-ack** voice-port configuration command. To disable the acknowledgment, use the **no** form of this command.

disconnect-ack
no disconnect-ack

Syntax Description

This command has no arguments or keywords.

Defaults

FXS voice ports return an acknowledgment upon receipt of a disconnect signal.

Command Mode

Voice-port configuration

Command History

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

Usage Guidelines

This command configures an FXS voice port to remove line power if the equipment on an FXS loop-start trunk disconnects first.

Examples

The following example turns off the disconnect acknowledgment signal on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
router(config-voiceport)# no disconnect-ack
```

The following example turns off the disconnect acknowledgment signal on voice port 1/1/0 on a Cisco 2600 or 3600:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# no disconnect-ack
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.

idle-voltage

To specify the idle voltage on an FXS voice port, use the **idle-voltage** voice-port configuration command. Use the **no** form of this command to restore the default idle voltage.

idle-voltage {**high** | **low**}
no idle-voltage

Syntax Description

high	The talk-battery (tip-to-ring) voltage is high (–48V) when the FXS port is idle.
low	The talk-battery (tip-to-ring) voltage is low (–24V) when the FXS port is idle

Defaults

The idle voltage is –24V.

Command Mode

Voice-port configuration

Command History

Release	Modification
12.0(4)T	This command was introduced on the Cisco MC3810 series.

Usage Guidelines

The **idle-voltage** command applies only to FXS voice ports on Cisco MC3810 series concentrators.

Some fax equipment and answering machines require a –48V idle voltage to be able to detect an off-hook condition in a parallel phone.

If the idle voltage is setting is **high**, the talk battery reverts to –24V whenever the voice port is active (off hook).

Examples

The following example sets the idle voltage to –48V on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
   router(config-voiceport)# idle-voltage high
```

The following example restores the default idle voltage (–24V) on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
   router(config-voiceport)# no idle-voltage
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.

ignore

To configure the E&M or E&M MELCAS voice port to ignore specific receive bits, use the **ignore** voice-port configuration command. Use the **no** form of this command to restore the default value.

ignore {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}
no ignore {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}

Syntax Description

rx-a-bit	Ignores the receive A bit.
rx-b-bit	Ignores the receive B bit.
rx-c-bit	Ignores the receive C bit.
rx-d-bit	Ignores the receive D bit.

Defaults

The default is mode-dependent:

E&M:

no ignore rx-a-bit
ignore rx-b-bit, rx-c-bit, rx-d-bit

E&M MELCAS:

no ignore rx-b-bit, rx-c-bit, rx-d-bit

Command Mode

Voice-port configuration

Command History

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

Usage Guidelines

This command applies to E&M digital voice ports associated with T1/E1 controllers. Repeat the command for each receive bit to be configured. Use this command with the **define** command.

Examples

To configure voice-port 1/1 on a Cisco MC3810 to ignore receive bits a, b, and c and to monitor receive bit d, enter the following commands:

```
router(config)# voice-port 1/1
router(config-voiceport)# ignore rx-a-bit
router(config-voiceport)# ignore rx-b-bit
router(config-voiceport)# ignore rx-c-bit
router(config-voiceport)# no ignore rx-d-bit
```

To configure voice-port 1/0/0 on a Cisco 3600 to ignore receive bits a, c, and d and to monitor receive bit b, enter the following commands:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# ignore rx-a-bit
router(config-voiceport)# ignore rx-c-bit
router(config-voiceport)# ignore rx-d-bit
router(config-voiceport)# no ignore rx-b-bit
```

Related Commands

Command	Description
condition	Manipulates the signaling bit-pattern for all voice signaling types.
define	Defines the transmit and receive bits for E&M and E&M MELCAS voice signaling.
show voice port	Displays configuration information for voice ports.

loss-plan

To specify the analog-to-digital gain offset for an analog FXO or FXS voice port, enter the **codec** dial-peer configuration command. Use the **no** form of this command to restore the default value.

loss-plan {**plan1** | **plan2** | **plan3** | **plan4** | **plan5** | **plan6** | **plan7** | **plan8** | **plan9**}
no loss-plan

Syntax Description

plan1	FXO: A-D gain = 0 dB, D-A gain = 0 dB FXS: A-D gain = –3 dB, D-A gain = –3 dB
plan2	FXO: A-D gain = 3 dB, D-A gain = 0 dB FXS: A-D gain = 0 dB, D-A gain = –3 dB
plan3	FXO: A-D gain = –3 dB, D-A gain = 0 dB FXS: Not applicable
plan4	FXO: A-D gain = –3 dB, D-A gain = –3 dB FXS: Not applicable
plan5	FXO: Not applicable FXS: A-D gain = –3 dB, D-A gain = –10 dB
plan6	FXO: Not applicable FXS: A-D gain = 0 dB, D-A gain = –7 dB
plan7	FXO: A-D gain = 7 dB, D-A gain = 0 dB FXS: A-D gain = 0 dB, D-A gain = –6 dB
plan8	FXO: A-D gain = 5 dB, D-A gain = –2 dB FXS: Not applicable
plan9	FXO: A-D gain = 6 dB, D-A gain = 0 dB FXS: Not applicable

Defaults

FXO: A-D gain = 0 dB, D-A gain = 0 dB (loss plan 1)

FXS: A-D gain = –3 dB, D-A gain = –3 dB (loss plan 1)

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	The following additional signal level choices were added: plan 3, plan 4, plan 8, and plan 9.

Usage Guidelines

This command sets the analog signal level difference (offset) between the analog voice port and the digital signal processor (DSP). Each loss plan specifies a level offset in both directions—from the analog voice port to the DSP (A-D) and from the DSP to the analog voice port (D-A).

Use this command to obtain the required levels of analog voice signals to and from the DSP.

This command is supported only on Cisco MC3810 series concentrators, on FXO and FXS analog voice ports.

Example

The following example configures FXO voice port 1/6 for a –3 dB offset from the voice port to the DSP and a 0 dB offset from the DSP to the voice port:

```
router(config)# voice-port 1/6
router(config-voiceport)# loss-plan plan3
```

The following example configures FXS voice port 1/1 for a 0 dB offset from the voice port to the DSP and a –7 dB offset from the DSP to the voice port:

```
router(config)# voice-port 1/1
router(config-voiceport)# loss-plan plan6
```

Related Commands

Command	Description
impedance	Specifies the terminating impedance of the voice port interface. Used on FXO voice ports in correcting input levels.
input gain	Specifies the gain applied by a voice port to the input signal from the PBX or other customer premises equipment.
output attenuation	Specifies the attenuation applied by a voice port to the output signal toward the PBX or other customer premises equipment.

playout delay

To tune the playout buffer to accommodate packet jitter caused by switches in the WAN, use the **playout-delay** voice-port configuration command. Use the **no** form of this command to restore the default value.

playout-delay {**maximum** | **nominal**} *milliseconds*

no playout-delay {**maximum** | **nominal**}

Syntax Description

maximum	The delay time the DSP allows before starting to discard voice packets. The default is 160 milliseconds.
nominal	The initial (and minimum allowed) delay time the DSP inserts before playing out voice packets. The default is 80 milliseconds
milliseconds	Playout-delay value in milliseconds. The range for maximum playout delay is 40 to 320, and the range for nominal playout delay is 40 to 240.

Defaults

The default maximum delay is 160 milliseconds.
The default nominal delay is 80 milliseconds.

Command Mode

Voice-port configuration

Command History

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

Usage Guidelines

If there is excessive break-up of voice due to jitter with the default playout delay settings, increase the delay times. If your network is small and jitter is minimal, decrease the delay times to reduce delay.

Examples

The following example configures a nominal playout delay of 80 milliseconds and a maximum playout delay of 160 milliseconds on voice-port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
router(config-voiceport)# playout-delay nominal 80
router(config-voiceport)# playout-delay maximum 160
```

The following example configures a nominal playout delay of 80 milliseconds and a maximum playout delay of 160 milliseconds on voice-port 1/0/0 on the Cisco 2600 or 3600:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# playout-delay nominal 80
router(config-voiceport)# playout-delay maximum 160
```

Related Commands

Command	Description
vad	Enables voice activity detection.

ring cadence

To specify the ring cadence for an FXS voice port, use the **ring cadence** voice-port configuration command. Use the **no** form of this command to restore the default value.

ring cadence {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] [**define** *pulse interval*]}
no ring cadence

Syntax Description

pattern01	2 seconds on, 4 seconds off
pattern02	1 second on, 4 seconds off
pattern03	1.5 seconds on, 3.5 seconds off
pattern04	1 second on, 2 seconds off
pattern05	1 second on, 5 seconds off
pattern06	1 second on, 3 seconds off
pattern07	0.8 second on, 3.2 seconds off
pattern08	1.5 seconds on, 3 seconds off
pattern09	1.2 seconds on, 3.7 seconds off
pattern09	1.2 seconds on, 4.7 seconds off
pattern11	0.4 second on, 0.2 second off, 0.4 second on, 2 seconds off
pattern12	0.4 second on, 0.2 second off, 0.4 second on, 2.6 seconds off
define	User-definable ring cadence pattern. Each number pair specifies one ring-pulse time and one ring-interval time. You must enter numbers in pairs, and you can enter 1 to 6 pairs. The second number in the last pair that you enter specifies the interval between rings.
pulse	A number (1 or 2 digits) specifying ring pulse (on) time in hundreds of milliseconds. The range is 1 to 50, for pulses of 100 ms to 5000 ms. For example: 1 = 100 ms; 10 = 1 s, 40 = 4 s.
interval	A number (1 or 2 digits) specifying ring interval (off) time in hundreds of milliseconds. The range is 1 to 50, for pulses of 100 to 5000 ms. For example: 1 = 100 ms; 10 = 1 s, 40 = 4 s.

Defaults

Ring cadence defaults to the pattern you specify with the **cptone** command.

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers, and the **patternXX** syntax was introduced.

Usage Guidelines

The **patternXX** keyword provides preset ring cadence patterns for use on any platform. The **define** keyword allows you to create a custom ring cadence. On the Cisco 2600 and 3600 series routers, only one or two pairs of digits can be entered under the **define** keyword.

Examples

The following example configures the ring cadence for 1 second on and 4 seconds off on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
router(config-voiceport)# ring cadence pattern02
```

The following example configures the ring cadence for 1 second on, 1 second off, 1 second on, and 5 seconds off on voice port 1/2 on a Cisco MC3810:

```
voice-port 1/2
router(config-voiceport)# ring cadence define 10 10 10 50
```

The following example configures the ring cadence for 1 second on and 2 seconds off on voice port 1/0/0 on a Cisco 2600 or 3600:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# ring cadence pattern04
```

Related Commands

Command	Description
ring frequency	Specifies the ring frequency for an FXS voice port.
cptone	Specifies the default tone, ring, and cadence settings according to country.

show call active voice

To show the active call table, use the **show call active voice** EXEC command.

show call active voice

Syntax Description

This command has no arguments or keywords.

Command Mode

User EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 2600 and 3600.
12.0(3)XG	Support for VoFR was added.
12.0(4)T	This command was first supported on the Cisco 7200 series.
12.0(7)XK	This command was first supported on the Cisco MC3810 series.

Usage Guidelines

This command applies to Voice over IP, Voice over Frame Relay, and Voice over ATM on the Cisco 2600, 3600, and MC3810 series.

Use this command to display the contents of the active call table, which shows all of the calls currently connected through the router. This command displays information about call times, dial peers, connections, Quality of Service, and other status and statistical information.

See Table 1 for a listing of the information types associated with this command.

Example

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

```
router# show call active voice
GENERIC: SetupTime=21072 Index=0 PeerAddress= PeerSubAddress= PeerId=0
PeerIfIndex=0 LogicalIfIndex=0 ConnectTime=0 CallState=3 CallOrigin=2 ChargedUnits=0
InfoType=0 TransmitPackets=375413 TransmitBytes=7508260 ReceivePackets=377734
ReceiveBytes=7554680

VOIP: ConnectionId[0x19BDF910 0xAF500007 0x0 0x58ED0] RemoteIPAddress=17635075
RemoteUDPPort=16394 RoundTripDelay=0 SelectedQoS=0 SessionProtocol=1
SessionTarget= OnTimeRvPlayout=0 GapFillWithSilence=0 GapFillWithPrediction=600
```

```
GapFillWithInterpolation=0 GapFillWithRedundancy=0 HiWaterPlayoutDelay=110
LoWaterPlayoutDelay=64 ReceiveDelay=94 VADEnable=0 CoderTypeRate=0

GENERIC: SetupTime=21072 Index=1 PeerAddress=+14085271001 PeerSubAddress=
PeerId=0 PeerIfIndex=0 LogicalIfIndex=5 ConnectTime=21115 CallState=4 CallOrigin=1
ChargedUnits=0 InfoType=1 TransmitPackets=377915 TransmitBytes=7558300
ReceivePackets=375594 ReceiveBytes=7511880

TELE: ConnectionId=[0x19BDF910 0xAF500007 0x0 0x58ED0] TxDuration=16640
VoiceTxDuration=16640 FaxTxDuration=0 CoderTypeRate=0 NoiseLevel=0 ACOMLevel=4
OutSignalLevel=-440 InSignalLevel=-440 InfoActivity=2 ERLLevel=227
SessionTarget=
```

Table 1 provides an alphabetical listing of the fields in this output and a description of each field.

Table 1 Show Call Active Voice Field Descriptions

Field	Description
ACOM Level	Current ACOM level for the call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CallState	Current state of the call.
CoderTypeRate	Negotiated coder transmit rate of voice/fax compression during the call.
ConnectionId	Global call identifier of a gateway call.
ConnectTime	Time at which the call was connected.
Dial-Peer	Tag of the dial peer transmitting this call.
ERLLevel	Current Echo Return Loss (ERL) level for this call.
FaxTxDuration	Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithSilence	Duration of voice signal replaced with silence because voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with 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. An example of such pullout is frame-eraser or frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received on time from voice gateway for this call.
GapFillWithRedundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because voice data was lost or not received on time from voice gateway 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 the call.

Table 1 Show Call Active Voice Field Descriptions (Continued)

Field	Description
NoiseLevel	Active noise level for the call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
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 the voice call.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system 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 the call.
SelectedQoS	Selected RSVP quality of service (QoS) for the 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 the 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 the call.
TransmitPackets	Number of packets transmitted from this peer during the call.
TxDuration	Duration of transmit path open from this peer to the voice gateway for the call.
VADEnable	Whether or not voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

Related Commands

Command	Description
show call history voice	Displays the call history table.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays the number expansions configured.
show voice port	Displays configuration information about a specific voice port.

show call history voice

To display the call history table, use the **show call history voice** EXEC command.

show call history voice [**last** *number* | **brief**]

Syntax Description

last *number*	(Optional) Displays the last calls connected, where the number of calls displayed is defined by the argument *number.* Valid entries for the argument *number* are numbers from 1 to 2147483647.
brief	(Optional) Displays abbreviated call history information for each leg of a call.

Command Mode

User EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600.
12.0(3)XG	Support for VoFR was added.
12.0(4)T	The **brief** keyword was added and the command was first supported on the Cisco 7200 series.
12.0(7)XK	Support for **brief** the keyword was added on the Cisco MC3810 platform.

Usage Guidelines

This command applies to all voice applications on the Cisco 2600, 3600, MC3810, and 7200 platforms.

Use the **show call history voice** privileged EXEC command to display the call history table. The call history table contains a listing of all voice calls connected through this router in descending time order. You can display subsets of the call history table by using specific keywords. To display the last calls connected through this router, use the keyword **last**, and define the number of calls to be displayed with the argument *number*. To display a shortened version of the call history table, use the keyword **brief**.

Example

The following is sample output from the **show call history voice** command for a VoFR call using the frf11-trunk session protocol:

```
router# show call history voice last 1
   GENERIC:
   SetupTime=8283963 ms
   Index=3149
   PeerAddress=3623110
   PeerSubAddress=
   PeerId=3400
   PeerIfIndex=18
   LogicalIfIndex=0
   DisconnectCause=3F
   DisconnectText=service or option not available, unspecified
   ConnectTime=8283963
   DisconectTime=8285463
   CallOrigin=1
   ChargedUnits=0
   InfoType=2
   TransmitPackets=94
   TransmitBytes=2751
   ReceivePackets=0
   ReceiveBytes=0

   VOFR:
   ConnectionId=[0x3D4B232D 0x6A900627 0x0 0x4F00852]
   Subchannel=[Interface Serial0/0, DLCI 160, CID 10]
   SessionProtocol=frf11-trunk
   SessionTarget=Serial0/0 160 10
   CalledNumber=2603100
   VADEnable=ENABLED
   CoderTypeRate=g729r8
   CodecBytes=30
   SignalingType=cas
   DTMFRelay=DISABLED
   UseVoiceSequenceNumbers=DISABLED

   GENERIC:
   SetupTime=8283963 ms
   Index=3150
   PeerAddress=2601100
   PeerSubAddress=
   PeerId=1100
   PeerIfIndex=7
   LogicalIfIndex=0
   DisconnectCause=3F
   DisconnectText=service or option not available, unspecified
   ConnectTime=8283964
   DisconectTime=8285464
   CallOrigin=2
   ChargedUnits=0
   InfoType=2
   TransmitPackets=0
   TransmitBytes=-121
   ReceivePackets=94
   ReceiveBytes=2563
   TELE:
   ConnectionId=[0x3D4B232D 0x6A900627 0x0 0x4F00852]
   TxDuration=15000 ms
   VoiceTxDuration=2010 ms
   FaxTxDuration=0 ms
   CoderTypeRate=g729r8
   NoiseLevel=-68
```

```
ACOMLevel=20
SessionTarget=
```

The following is sample output from the **show call history voice** command for a VoIP call:

```
router# show call history voice
   GENERIC:
   SetupTime=20405
   Index=0
   PeerAddress=
   PeerSubAddress=
   PeerId=0
   PeerIfIndex=0
   LogicalIfIndex=0
   DisconnectCause=NORMAL
   DisconnectText=
   ConnectTime=0
   DisconectTime=20595
   CallOrigin=2
   ChargedUnits=0
   InfoType=0
   TransmitPackets=0
   TransmitBytes=0
   ReceivePackets=0
   ReceiveBytes=0

   VOIP:
   ConnectionId[0x19BDF910 0xAF500006 0x0 0x56590]
   RemoteIPAddress=17635075
   RemoteUDPPort=16392
   RoundTripDelay=0
   SelectedQoS=0
   SessionProtocol=1
   SessionTarget=
   OnTimeRvPlayout=0
   GapFillWithSilence=0
   GapFillWithPrediction=0
   GapFillWithInterpolation=0
   GapFillWithRedundancy=0
   HiWaterPlayoutDelay=0
   LoWaterPlayoutDelay=0
   ReceiveDelay=0
   VADEnable=0
   CoderTypeRate=0

   TELE: ConnectionId=[0x19BDF910 0xAF500006 0x0 0x56590]
   TxDuration=3030
   VoiceTxDuration=2700
   FaxTxDuration=0
   CoderTypeRate=0
   NoiseLevel=0
   ACOMLevel=0
   SessionTarget=
```

Table 2 provides an alphabetical listing of the fields in this output and a description of each field.

Table 2 Show Call History Voice Field Descriptions

Field	Description
ACOMLevel	Average ACOM level for this call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CoderTypeRate	Negotiated coder rate. This value specifies the transmit rate of voice/fax compression to its associated call leg for the call.
ConnectionID	Global call identifier for the gateway call.
ConnectTime	Time the call was connected.
DisconnectCause	Description explaining why the call was disconnected.
DisconnectText	Descriptive text explaining the disconnect reason.
DisconnectTime	Time the call was disconnected.
FaxDuration	Duration of fax transmitted from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing this value by the TxDuration value.
GapFillWithSilence	Duration of voice signal replaced with silence because the voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.
GapFillWithRedundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because the voice data was lost or not received on time from the voice gateway for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during the voice call.
Index	Index number identifying the voice-peer for this call.
InfoType	Information type for this call.
LogicalIfIndex	Index of the logical voice port for this call.
LoWaterPlayoutDelay	Low water mark Voice Playout FIFO Delay during the voice call.
NoiseLevel	Average noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
PeerAddress	Destination pattern or number to which this call is connected.
PeerId	ID value of the peer entry table to which this call was made.
PeerIfIndex	Index number of the logical interface through which this call was made. For ISDN media, this would be the index number of the B channel used for the call.
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 the voice call.
ReceivePackets	Number of packets received by this peer during the call.

Table 2 Show Call History Voice Field Descriptions (Continued)

Field	Description
RemoteIPAddress	Remote system IP address for the call.
RemoteUDPPort	Remote system UDP listener port to which voice packets for this call are transmitted.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone for this call.
SelectedQoS	Selected RSVP quality of service for the call.
SessionProtocol	Session protocol to be used for an Internet call between the local and remote router via the IP backbone.
SessionTarget	Session target of the peer used for the call.
SetUpTime	Value of the System UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted by this peer during the call.
TransmitPackets	Number of packets transmitted by this peer during the call.
TxDuration	Duration of the transmit path open from this peer to the voice gateway for the call.
VADEnable	Whether or not voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmitted from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration by the TxDuration value.

Related Commands

Command	Description
show call active voice	Displays the contents of the active call table.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays the number expansions configured.
show voice port	Displays configuration information about a specific voice port.

show voice call

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

For the Cisco 2600 and 3600 series with analog voice ports:

show voice call [*slot/subunit/port* | **summary**]

For the Cisco 2600 and 3600 series with digital voice ports (with T1 packet voice trunk network modules):

show voice call [*slot/port:ds0-group* | **summary**]

For the Cisco MC3810 series with analog voice ports:

show voice call [*slot/port* | **summary**]

For the Cisco MC3810 series with digital voice ports:

show voice call [*slot:ds0-group* | **summary**]

Syntax Description

summary	(Optional) Show a summary of the call status, not the detailed report.
voice-port	(Optional) Displays the call status for a specified voice port.

Command Mode

User EXEC

Command History

Release	Modification
11.3 MA	This command was introduced for the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

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.

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

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

- Call processing state machine
- Protocol state machine

Sample Display

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:

```
router# show voice call summary

PORT    CODECE   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 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:

```
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 is a sample display 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
```

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	Shows the current status of all DSP voice channels.
show voice port	Displays configuration information about a specific voice port.

show voice dsp

To show the configuration status for all configured DSP voice channels on the Cisco router or concentrator, use the **show voice dsp** EXEC command.

show voice dsp

Syntax Description

This command has no arguments or keywords.

Command Mode

User EXEC

Command History

Release	Modification
11.3 MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600, and the display format was modified.

Usage Guidelines

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

Use this command when abnormal behavior in the DSP voice channels occurs.

Sample Display

The following is a sample display from the **show voice dsp** command on a Cisco MC3810:

```
Router#show voice dsp
                                BOOT                          PAK
TYPE DSP CH CODEC    VERS STATE STATE   RST AI PORT    TS ABORT   TX/RX-PAK-CNT
==== === == ======== ==== ===== ======= === == ======= == ===== ===============
C549 001 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 002 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 003 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 004 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 005 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 006 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
```

Table 3 provides an alphabetical listing of the fields in this output and a description of each field.

Table 3 Show Voice DSP Field Descriptions

Field	Description
AI	Number of alarm indications received from the DSP, which may point to abnormality of DSP firmware.
BOOT STATE	Applicable to Cisco MC3810 only of dynamic reload of DSP is permitted.
CH	Voice channel number in DSP.
CODEC	Cisco MC3810 with HCM and Cisco 2600 and 3600 digital: If the DSP channel is in use, this indicates what codec it is using. If a DSP channel is not in use, this indicates the complexity level configured. Cisco MC3810 with VCM and Cisco 2600 and 3600 analog: Indicates what codec is loaded.
DSP	DSP number.
PAK ABORT	The number of DSP packets dropped due to DSP failure in picking up packets from the host.
PORT	The port number associated with the DSP channel. This is a fixed port number on the Cisco 2600 and 3600; this number may change with each new call on the Cisco MC3810.
RST	The number of DSP resets since the most recent **clear counters** entry.
STATE	The busy/idle state of the DSP channel.
TS	The backplane timeslot associated with this DSP channel. This is a fixed timeslot on the Cisco 2600 and 3600; this number may change with each new call on the Cisco MC3810.
TX/RX-PAK-CNT	An ordered pair of transmit and receive packet counts processed by the DSP since the previous **clear counters** command was entered.
TYPE	DSP hardware type.
VERS	Version and revision of DSP hardware, in X,Y format.

Related Commands

Command	Description
clear counters	Clears all the current interface counters from the interface.
show voice port	Displays configuration information about a specific voice port.

show voice port

To display configuration information about a specific voice port, use the **show voice port** EXEC command.

For the Cisco 2600 and 3600 series with analog voice ports:

show voice port [*slot/subunit/port* | **summary**]

For the Cisco 2600 and 3600 series with digital voice ports (with T1 packet voice trunk network modules):

show voice port [*slot/port:ds0-group* | **summary**]

For the Cisco MC3810 series with analog voice ports:

show voice port [*slot/port* | **summary**]

For the Cisco MC3810 series with digital voice ports:

show voice port [*slot:ds0-group* | **summary**]

Syntax Description

For the Cisco 2600 and 3600 series with analog voice ports:

slot/subunit/port	(Optional) Displays information for the analog voice port 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 voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) *port* specifies an analog voice port number. Valid entries are 0 and 1.
summary	(Optional) Displays a summary of all voice ports.

For the Cisco 2600 and 3600 series with digital voice ports:

slot/port:ds0-group	(Optional) Displays information for the digital voice port you specify with the *slot/port:ds0-group* designation. *slot* 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. *port* 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.) *ds0-group* 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.

For the Cisco MC3810 series with analog voice ports:

slot/port	(Optional) Displays information for the analog voice port you specify with the *slot/port* designation. *slot* 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. *port* specifies an analog voice port number. Valid entries are 1 to 6.
summary	(Optional) Displays a summary of all voice ports.

For the Cisco MC3810 series with digital voice ports:

slot:ds0-group	(Optional) Displays information for the digital voice port you specify with the *slot:ds0-group* designation. *slot* specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). *ds0-group* 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.

Command Mode

User EXEC

Command History

Release	Modification
11.3(1) T	This command was introduced.
12.0(5)XK and 12.0(6)T	The *ds0-group* argument was added for the Cisco 2600 and 3600 series routers.
12.0(7)XK	The **summary** keyword was added for the Cisco 2600 and 3600 series routers. The *ds0-group* argument was added for the Cisco MC3810.

Usage Guidelines

Use the **show voice port** privileged EXEC command to display configuration and voice-interface-card-specific information about a specific port.

Example

The following is sample output from the **show voice port summary** command for all voice ports on a Cisco MC3810 with an analog voice module (AVM):

```
router# show voice port summary
IN  OUT   ECHO
PORT SIG-TYPE     ADMIN OPER IN-STATUS OUT-STATUS GAIN ATTN CANCEL
1/1  fxs-ls       up    up   on-hook    idle        0    0    y
1/2  fxs-ls       up    up   on-hook    idle        0    0    y
1/3  e&m-wnk      up    up   idle       idle        0    0    y
1/4  e&m-wnk      up    up   idle       idle        0    0    y
1/5  fxo-ls       up    up   idle       on-hook     0    0    y
1/6  fxo-ls       up    up   idle       on-hook     0    0    y
```

The following is sample output from the **show voice port summary** command on a Cisco MC3810 with a digital voice module (DVM):

```
                                  IN       OUT
PORT   CH SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
====== == ========== ===== ==== ======== ======== ==
0:17   18 fxo-ls     down  down idle     on-hook  y
0:18   19 fxo-ls     up    dorm idle     on-hook  y
0:19   20 fxo-ls     up    dorm idle     on-hook  y
0:20   21 fxo-ls     up    dorm idle     on-hook  y
0:21   22 fxo-ls     up    dorm idle     on-hook  y
0:22   23 fxo-ls     up    dorm idle     on-hook  y
0:23   24 e&m-imd    up    dorm idle     idle     y
1/1    -- fxs-ls     up    dorm on-hook  idle     y
1/2    -- fxs-ls     up    dorm on-hook  idle     y
1/3    -- e&m-imd    up    dorm idle     idle     y
1/4    -- e&m-imd    up    dorm idle     idle     y
1/5    -- fxo-ls     up    dorm idle     on-hook  y
1/6    -- fxo-ls     up    dorm idle     on-hook  y
Elements :
sys/voip/ccvpm                 vpm_htsp.c (107)
sys/voip/ccvtsp                vtsp_core.c (167)
sys/voip/cli                   voiceport_action.c (58)
```

The following is sample output from the **show voice port** command for an E&M analog voice port on a Cisco 3600:

```
router# show voice port 1/0/0
E&M Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is E&M
 Operation State is unknown
 Administrative State is unknown
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is disabled
 Non Linear Processing is disabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is disabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 0 s
 Interdigit Time Out is set to 0 s
 Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
```

```
 Voice card specific Info Follows:
 Signal Type is wink-start
 Operation Type is 2-wire
 Impedance is set to 600r Ohm
 E&M Type is unknown
 Dial Type is dtmf
 In Seizure is inactive
 Out Seizure is inactive
 Digit Duration Timing is set to 0 ms
 InterDigit Duration Timing is set to 0 ms
 Pulse Rate Timing is set to 0 pulses/second
 InterDigit Pulse Duration Timing is set to 0 ms
 Clear Wait Duration Timing is set to 0 ms
 Wink Wait Duration Timing is set to 0 ms
 Wink Duration Timing is set to 0 ms
 Delay Start Timing is set to 0 ms
 Delay Duration Timing is set to 0 ms
```

The following is sample output from the **show voice port** command for an FXS analog voice port on a Cisco 3600:

```
router# show voice port 1/0/0
Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is FXS
 Operation State is DORMANT
 Administrative State is UP
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 25 Hz
 Hook Status is On Hook
 Ring Active Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is inactive
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Hook Flash Duration Timing is set to 600 ms
```

The following is sample output from the **show voice port** command for an FXS analog voice port on a Cisco MC3810:

```
router# show voice port 1/2
Voice port 1/2 Slot is 1, Port is 2
 Type of VoicePort is FXS
 Operation State is UP
 Administrative State is UP
 No Interface Down Failure
```

```
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Coder Type is g729ar8
 Companding Type is u-law
 Voice Activity Detection is disabled
 Ringing Time Out is 180 s
 Wait Release Time Out is 30 s
 Nominal Playout Delay is 80 milliseconds
 Maximum Playout Delay is 160 milliseconds

 Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing Voice
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Impedance is set to 600r Ohm
 Analog interface A-D gain offset = -3 dB
 Analog interface D-A gain offset = -3 dB
 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 20 Hz
 Hook Status is On Hook
 Ring Active Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is active
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Ring Cadence are [20 40] * 100 msec
 InterDigit Pulse Duration Timing is set to 500 ms
```

The following is sample output from the **show voice port** command for an E&M digital voice port on a Cisco 3600:

```
router# show voice port 1/0:1

receEive and transMit Slot is 1, Sub-unit is 0, Port is 1
 Type of VoicePort is E&M
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Region Tone is set for US
```

Table 4 explains the fields in the sample output.

Table 4 Show Voice Port Field Descriptions

Field	Description
Administrative State	Administrative state of the voice port.
Alias	User-supplied alias for this voice port.
Analog interface A-D gain offset	Offset of the gain for analog-to-digital conversion.
Analog interface D-A gain offset	Offset of the gain for digital-to-analog conversion.
Clear Wait Duration Timing	Time of inactive seizure signal to declare call cleared.
Coder Type	Voice compression mode used.
Companding Type	Companding standard used to convert between analog and digital signals in PCM systems.
Connection Mode	Connection mode of the interface.
Connection Number	Full E.164 telephone number used to establish a connection with the trunk or PLAR mode.
Currently Processing	Type of call currently being processed: none, voice, or fax.
Delay Duration Timing	Maximum delay signal duration for delay dial signaling.
Delay Start Timing	Timing of generation of delayed start signal from detection of incoming seizure.
Description	Description of the voice port.
Dial Type	Out-dialing type of the voice port.
Digit Duration Timing	DTMF Digit duration in milliseconds.
E&M Type	Type of E&M interface.
Echo Cancel Coverage	Echo Cancel Coverage for this port.
Echo Cancellation	Whether or not echo cancellation is enabled for this port.
Hook Flash Duration Timing	Maximum length of hook flash signal.
Hook Status	Hook status of the FXO/FXS interface.
Impedance	Configured terminating impedance for the E&M interface.
In Gain	Amount of gain inserted at the receiver side of the interface.
In Seizure	Incoming seizure state of the E&M interface.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
InterDigit Duration Timing	DTMF interdigit duration in milliseconds.
InterDigit Pulse Duration Timing	Pulse dialing interdigit timing in milliseconds.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Maintenance Mode	Maintenance mode of the voice port.
Maximum Playout Delay	The amount of time before the Cisco MC3810 DSP starts to discard voice packets from the DSP buffer.
Music On Hold Threshold	Configured Music-On-Hold Threshold value for this interface.
Noise Regeneration	Whether or not background noise should be played to fill silent gaps if VAD is activated.
Nominal Playout Delay	The amount of time the Cisco MC3810 DSP waits before starting to play out the voice packets from the DSP buffer.

Table 4 Show Voice Port Field Descriptions (Continued)

Field	Description
Non-Linear Processing	Whether or not non-linear processing is enabled for this port.
Number of signaling protocol errors	Number of signaling protocol errors.
Operations State	Operation state of the port.
Operation Type	Operation of the E&M signal: two-wire or four-wire.
Out Attenuation	Amount of attenuation inserted at the transmit side of the interface.
Out Seizure	Outgoing seizure state of the E&M interface.
Port	Port number for this interface associated with the voice interface card.
Pulse Rate Timing	Pulse dialing rate in pulses per second (pps).
Region Tone	Configured regional tone for this interface.
Ring Active Status	Ring active indication.
Ring Cadence	Configured ring cadence for this interface.
Ring Frequency	Configured ring frequency for this interface.
Ring Ground Status	Ring ground indication.
Ringing Time Out	Ringing time out duration.
Signal Type	Type of signaling for a voice port: loop-start, ground-start, wink-start, immediate, and delay-dial.
Slot	Slot used in the voice interface card for this port.
Sub-unit	Subunit used in the voice interface card for this port.
Tip Ground Status	Tip ground indication.
Type of VoicePort	Type of voice port: FXO, FXS, and E&M.
The Interface Down Failure Cause	Text string describing why the interface is down,
Voice Activity Detection	Whether Voice Activity Detection is enabled or disabled.
Wait Release Time Out	The time a voice port stays in the call-failure state while the Cisco MC3810 sends a busy tone, reorder tone, or an out-of-service tone to the port.
Wink Duration Timing	Maximum wink duration for wink start signaling.
Wink Wait Duration Timing	Maximum wink wait duration for wink start signaling.

Related Commands

Command	Description
show voice call	Displays the call status for all voice ports on the Cisco router or concentrator.
show call history voice	Displays the call history table.
show dial-peer voice	Displays configuration information about dial peers.
show num-exp	Displays the number expansions that are configured.

timeouts ringing

To configure the timeout value for ringing, use the **timeouts ringing** voice-port configuration command. Use the **no** form of this command to restore the default value.

timeouts ringing {*seconds* | **infinity**}
no timeouts ringing

Syntax Description

seconds	The duration in seconds that a voice port allows ringing to continue if a call is not answered. The range is 5 to 60000.
infinity	Ringing continues until the caller goes on hook.

Defaults

180 seconds

Command Mode

Voice-port configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

This command provides the capability to limit the length of time that a caller can continue ringing a telephone when there is no answer.

Examples

The following example configures voice port 1/1 on a Cisco MC3810 to allow ringing for 600 seconds:

```
router(config)# voice-port 1/1
router(config-voiceport)# timeouts ringing 600
```

The following example configures voice port 0/0/1 on a Cisco 3600 to allow ringing for 600 seconds:

```
router(config)# voice-port 0/0/1
router(config-voiceport)# timeouts ringing 600
```

Related Commands

Command	Description
timeouts initial	Configures the initial-digit timeout value for a voice port.
timeouts interdigit	Configures the interdigit timeout value for a voice port.

timeouts wait-release

To configure the delay timeout before the system starts the process for releasing voice ports, use the **timeouts wait-release** voice-port configuration command. Use the **no** form of this command to restore the default value.

timeouts wait-release {*seconds* | **infinity**}
no timeouts wait-release

Syntax Description

seconds	The duration in seconds that a voice port stays in the call-failure state while the Cisco router or concentrator sends a busy tone, reorder tone, or an out-of-service tone to the port. The range is 3 to 3600. The default is 30.
infinity	The voice port is never released as long as the call-failure state remains.

Defaults

30 seconds

Command Mode

Voice-port configuration

Command History

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

Usage Guidelines

Use this command to limit the time a voice port can be held in a call failure state. After the timeout, the release sequence is enabled.

You can also use this command for voice ports with FXS loop-start signaling, to specify the time allowed for a caller to hang up before the voice port goes into the parked state.

Examples

The following example configures voice port 1/1 on a Cisco MC3810 to stay in the call-failure state for 180 seconds while a busy tone, reorder tone, or out-of-service tone is sent to the voice port:

```
router(config)# voice-port 1/1
router(config-voiceport)# timeouts wait-release 180
```

The following example configures voice port 0/0/1 on a Cisco 3600 to stay in the call-failure state for 180 seconds while a busy tone, reorder tone, or out-of-service tone is sent to the voice port:

```
router(config)# voice-port 0/0/1
router(config-voiceport)# timeouts wait-release 180
```

Related Commands

Command	Description
timeouts initial	Configures the initial-digit timeout value for a voice port.
timeouts interdigit	Configures the interdigit timeout value for a voice port.

timing guard-out

To specify the guard-out duration of an FXO voice port, use the **timing guard-out** voice-port configuration command. Use the **no** form of this command to restore the default value.

timing guard-out *milliseconds*
no timing guard-out

Syntax Description

milliseconds	Duration in milliseconds of the guard-out period. The range is 300 to 3000. The default is 2000.

Defaults

2000 milliseconds

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1)MA5	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

This command applies to the Cisco 2600, 3600, and MC3810 platforms.

This command is supported on FXO voice ports only.

Examples

The following example configures the timing guard-out duration on a Cisco MC3810 voice port to 1000 milliseconds:

```
router(config)# voice-port 1/1
router(config-voiceport)# timing guard-out 1000
```

The following example configures the timing guard-out duration on a Cisco 2600 or 3600 voice port to 1000 milliseconds:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# timing guard-out 1000
```

timing percentbreak

To specify the percentage of the break period for dialing pulses for a voice port, use the **timing percentbreak** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing percentbreak *percent*
no timing percentbreak

Syntax Description

percent	Percentage of the break period for dialing pulses. Valid entries are numbers 20 to 80. The default is 50.

Defaults

50 percent

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1) MA4	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

This command is supported on FXO and E&M voice ports only.

Examples

The following example configures the break period percentage on a Cisco MC3810 voice port to 30 percent:

```
router(config)# voice-port 1/1
router(config-voiceport)# timing percentbreak 30
```

The following example configures the break period percentage on a Cisco 2600 or 3600 voice port to 30 percent:

```
router(config)# voice-port 0/0/1
router(config-voiceport)# timing percentbreak 30
```

Related Commands

Command	Description
timing pulse	Configures the pulse dialing rate for a voice port.
timing pulse-interdigit	Configures the pulse inter-digit timing for a voice port.

voice local-bypass

To configure local calls to bypass the digital signal processor (DSP), use the **voice local-bypass** global configuration command. Use the **no** form of this command to direct local calls through the DSP.

voice local-bypass
no voice local-bypass

Syntax Description

This command has no arguments or keywords.

Defaults

Local calls bypass the DSP.

Command Mode

Global configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

Local calls (calls between voice ports on a router or concentrator) normally bypass the DSP to minimize use of system resources. Use the **no** form of this command if you need to direct local calls through the DSP. Input gain and output attenuation can be configured only if calls are directed through the DSP.

Examples

The following example configures a Cisco MC3810, 2600, or 3600 to pass local calls through the DSP:

```
router(config)# no voice local-bypass
```

Related Commands

Command	Description
input gain	Configures receive gain value for a voice port.
output attenuation	Configures transmit attenuation value for a voice port.

voice vad-time

To change the minimum silence detection time for voice activity detection (VAD), use the **voice vad-time** global configuration command. Use the **no** form of this command to restore the default value.

voice vad-time *milliseconds*
no voice vad-time

Syntax Description

milliseconds	The waiting period in milliseconds before silence detection and suppression of voice-packet transmission. The range is 250 to 65536. The default is 250.

Defaults

250 milliseconds

Command Mode

Global configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco 2600, 3600, and MC3810.

Usage Guidelines

This command affects all voice ports on a router or concentrator, but it does not affect calls already in progress.

You can use this command in transparent CCS applications in which you want VAD to activate when the voice channel is idle, but not during active calls. With a longer silence detection delay, VAD reacts to the silence of an idle voice channel, but not to pauses in conversation.

This command does not affect voice codecs that have ITU-standardized built-in VAD features—for example, G.729B, G.729AB, G.723.1A. The VAD behavior and parameters of these codecs are defined exclusively by the applicable ITU standard.

Examples

The following example configures a 20-second delay before VAD silence detection is enabled:

```
router(config)# voice vad-time 20000
```

Related Commands

Command	Description
vad (dial peer)	Enables voice activity detection on a network dial peer.

Debug Commands

This section documents new or modified commands. All other commands used on these platforms are documented in the Cisco IOS Release 12.0 command reference publications.

- **debug vpm all**
- **debug vpm error**
- **debug vtsp all**
- **debug vtsp dsp**
- **debug vtsp port**
- **debug vtsp session**
- **debug vtsp stats**
- **debug vtsp vofr subframe**

The following debug commands have been removed in Cisco IOS Release 12.0(7)XK:

- **debug voice all**
- **debug voice cp**
- **debug voice eecm**
- **debug voice protocol**
- **debug voice signaling**
- **debug voice vofr**

debug vpm all

Use the **debug vpm all** command to enable all voice port module (VPM) debugging. Use the **no** form of this command to disable all VPM debugging.

debug vpm all
no debug vpm all

Syntax Description

This command has no arguments or keywords.

Defaults

VPM debugging is not enabled.

Command History

Release	Modification
11.3(1)T	This command was introduced for the Cisco 3600 series.
12.0(7)XK	This command was updated for the Cisco 2600, 3600, and MC3810.

Usage Guidelines

Use the **debug vpm all** command to enable the complete set of VPM debugging commands: **debug vpm dsp**, **debug vpm error**, **debug vpm port**, **debug vpm spi**, and **debug vpm trunk_sc**.

Execution of **no debug all** will turn off all port level debugging. It is usually a good idea to turn off all debugging and then enter the debug commands you are interested in one by one. This will help to avoid confusion about which ports you are actually debugging.

Examples

For sample outputs, refer to the individual commands in this chapter.

Related Commands

Command	Description
debug vpm port	Limits the **debug vpm all** command to a specified port.
show debug	Shows which debug commands are enabled.
debug vpm error	Enables DSP error tracing.
debug vtsp all	Enables the display of trunk conditioning supervisory component trace information.

debug vpm error

Use the **debug vpm error** command to enable DSP error tracing in voice port modules (VPMs). Use the **no** form of this command to disable DSP error tracing.

debug vpm error
no debug vpm error

Syntax Description

This command has no arguments or keywords.

Defaults

VPM debugging is not enabled.

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco 2600, 3600, and MC3810.

Usage Guidelines

Execution of **no debug all** will turn off all port level debugging. You should turn off all debugging and then enter the debug commands you are interested in one by one. This will help avoid confusion about which ports you are actually debugging.

Examples

The following example shows **debug vpm error** messages for Cisco 2600 or 3600 series router:

```
debug vpm error
```

The following example shows **debug vpm error** messages for a Cisco MC3810:

```
debug vpm error
```

The following example turns off **debug vpm error** debugging messages:

```
no debug vpm error
```

Related Commands

Command	Description
debug vpm all	Enables all VPM debugging.
debug vpm port	Limits the **debug vpm error** command to a specified port.
show debug	Shows which debug commands are enabled.

debug vtsp all

Use the **debug vtsp all** command to show debugging information for all of the **debug vtsp** commands. Use the **no** form of this command to disable debugging output.

debug vtsp all
no debug vtsp all

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp is not enabled.

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco 2600, 3600 and MC3810 platforms.

Usage Guidelines

The **debug vtsp all** command enables the following debug vtsp commands: **debug vtsp session**, **debug vtsp error**, and **debug vtsp dsp**. For more information or sample output, refer to the individual commands in this chapter.

Execution of **no debug vtsp all** will turn off all VTSP-level debugging. You should turn off all debugging and then enter the debug commands you are interested in one by one. This will help to avoid confusion about which ports you are actually debugging.

Related Commands

Command	Description
show debug	Shows which debug commands are enabled.
debug vtsp port	Limits vtsp debug output to a specific voice port.

debug vtsp dsp

Use the **debug vtsp dsp** command to show messages from the DSP to the access server. Use the **no** form of this command to disable debugging output.

debug vtsp dsp
no debug vtsp dsp

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp dsp is not enabled.

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco 2600, 3600, and MC3810 platforms.

Usage Guidelines

ON AS5300 ACCESS SERVERS

The **debug vtsp dsp** command shows messages from the DSP on the VFC to the router; this command can be useful if you suspect that the VFC is not functional. It is a simple way to check if the VFC is responding to off-hook indications.

ON 2600, 3600, MC3810 PLATFORMS

The **debug vtsp dsp** command shows messages from the DSP to the router.

Sample Display

The following example shows the collection of DTMF digits from the DSP on a Cisco AS5300 access server.

```
*Nov 30 00:44:34.491: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=3
*Nov 30 00:44:36.267: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=1
*Nov 30 00:44:36.571: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=0
*Nov 30 00:44:36.711: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=0
*Nov 30 00:44:37.147: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=2
```

Related Commands

Command	Description
debug vpm all	Enables all VPM debugging.
debug vtsp port	Limits vtsp debug output to a specific voice port.
show debug	Shows which debug commands are enabled.

debug vtsp error

Use the **debug vtsp error** command to display processing errors in the voice telephony service provider. Use the **no** form of this command to disable vtsp error debugging.

debug vtsp error
no debug vtsp error

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp errors is not enabled.

Command History

Release	Modification
12.0(7)XK	This command was first supported on the Cisco 2600, 3600 and MC3810 platforms.

Usage Guidelines

The **debug vtsp error** command can be used to check for mismatches in interface capabilities.

Sample Display

The following example shows sample output from the **debug vtsp error** command, in which a dialed number is not reachable because it is not configured.

```
router#deb vtsp error
Voice telephony call control error debugging is on

router#
*Mar  1 00:21:48.698:cc_api_call_setup_ind (vdbPtr=0x1575AB0,
callInfo={called=,called_oct3=0x81,calling=9999,calling_oct3=0x0,called_oct3a=0x0,
       fdest=0 peer_tag=1},callID=0x15896A4)
*Mar  1 00:21:48.698:cc_api_call_setup_ind type 3 , prot 0
*Mar  1 00:21:48.706:cc_process_call_setup_ind (event=0x16AD0E0) handed call to app
"SESSION"
*Mar  1 00:21:48.706:sess_appl:ev(23=CC_EV_CALL_SETUP_IND), cid(15), disp(0)
*Mar  1 00:21:48.706:sess_appl:ev(SSA_EV_CALL_SETUP_IND), cid(15), disp(0)
*Mar  1 00:21:48.706:ccCallSetContext (callID=0xF, context=0x1632898)
*Mar  1 00:21:48.706:ccCallSetupAck (callID=0xF)
*Mar  1 00:21:48.706:ccGenerateTone (callID=0xF tone=8)
*Mar  1 00:21:49.710:cc_api_call_digit_begin (vdbPtr=0x1575AB0, callID=0xF, digit=5,
flags=0x1, timestamp=0xB1AE6BC4, expiration=0x0)
*Mar  1 00:21:49.710:sess_appl:ev(10=CC_EV_CALL_DIGIT_BEGIN), cid(15), disp(0)
*Mar  1 00:21:49.710:cid(15)st(SSA_CS_MAPPING)ev(SSA_EV_DIGIT_BEGIN)
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
*Mar  1 00:21:49.714:ssaIgnore cid(15), st(SSA_CS_MAPPING),oldst(0), ev(10)
*Mar  1 00:21:49.778:cc_api_call_digit (vdbPtr=0x1575AB0, callID=0xF, digit=5,
duration=4165,tag 0, callparty 0 )
*Mar  1 00:21:49.778:sess_appl:ev(9=CC_EV_CALL_DIGIT), cid(15), disp(0)
*Mar  1 00:21:49.778:cid(15)st(SSA_CS_MAPPING)ev(SSA_EV_CALL_DIGIT)
```

```
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
*Mar  1 00:21:49.782:ssaDigit
*Mar  1 00:21:49.782:ssaDigit, callinfo , digit 5, tag 0,callparty 0
*Mar  1 00:21:49.782:ssaDigit, calling 9999,result 1
*Mar  1 00:21:49.915:cc_api_call_digit_begin (vdbPtr=0x1575AB0, callID=0xF, digit=5,
flags=0x1, timestamp=0xB1AF6B6C, expiration=0x0)
*Mar  1 00:21:49.915:sess_appl:ev(10=CC_EV_CALL_DIGIT_BEGIN), cid(15), disp(0)
*Mar  1 00:21:49.915:cid(15)st(SSA_CS_MAPPING)ev(SSA_EV_DIGIT_BEGIN)
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
*Mar  1 00:21:49.915:ssaIgnore cid(15), st(SSA_CS_MAPPING),oldst(0), ev(10)
*Mar  1 00:21:49.999:cc_api_call_digit (vdbPtr=0x1575AB0, callID=0xF, digit=5,
duration=95,tag 0, callparty 0 )
*Mar  1 00:21:49.999:sess_appl:ev(9=CC_EV_CALL_DIGIT), cid(15), disp(0)
*Mar  1 00:21:50.003:cid(15)st(SSA_CS_MAPPING)ev(SSA_EV_CALL_DIGIT)
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
*Mar  1 00:21:50.003:ssaDigit
*Mar  1 00:21:50.003:ssaDigit, callinfo , digit 55, tag 0,callparty 0
*Mar  1 00:21:50.003:ssaDigit, calling 9999,result -1
*Mar  1 00:21:50.003:ccCallDisconnect (callID=0xF, cause=0x1C tag=0x0)
*Mar  1 00:21:50.003:ccCallDisconnect (callID=0xF, cause=0x1C tag=0x0)
*Mar  1 00:21:50.007:vtsp_process_event():prev_state = 0.4 ,
 state = S_WAIT_RELEASE_NC, event = E_CC_DISCONNECT
 Invalid FSM  Input on channel 1/1:15
*Mar  1 00:21:52.927:vtsp_process_event():prev_state = 0.7 ,
 state = S_WAIT_RELEASE_RESP, event = E_TSP_CALL_FEATURE_IND
 Invalid FSM  Input on channel 1/1:15
*Mar  1 00:21:52.931:cc_api_call_disconnect_done(vdbPtr=0x1575AB0, callID=0xF, disp=0,
tag=0x0)
*Mar  1 00:21:52.931:sess_appl:ev(13=CC_EV_CALL_DISCONNECT_DONE), cid(15), disp(0)
*Mar  1 00:21:52.931:cid(15)st(SSA_CS_DISCONNECTING)ev(SSA_EV_CALL_DISCONNECT_DONE)
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
```

Related Commands

Command	Description
debug vpm all	Enables all VPM debugging.
debug vtsp port	Limits vtsp debug output to a specific voice port.
show debug	Shows which debug commands are enabled.

debug vtsp port

To observe the behavior of the VTSP state machine on a specific voice port, use the **debug vtsp port** command. Use the **no** form of the command to turn off the debug function.

For Cisco 2600 and 3600 series with analog voice ports:

debug vtsp port *slot/subunit/port*
no debug vtsp port *slot/subunit/port*

For Cisco 2600 and 3600 series with digital voice ports (with T1 packet voice trunk network modules):

debug vtsp port *slot/port:ds0-group*
no debug vtsp port *slot/port:ds0-group*

For Cisco MC3810 series with analog voice ports:

debug vtsp port *slot/port*
no debug vtsp port *slot/port*

For Cisco MC3810 series with digital voice ports:

debug vtsp port *slot/port*
no debug vtsp port *slot/ds0-group*

Syntax Description

For the Cisco 2600 and 3600 series with analog voice ports:

slot/subunit/port	Debugs the analog voice port 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 voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) *port* specifies an analog voice port number. Valid entries are 0 and 1.

For the Cisco 2600 and 3600 series with digital voice ports:

slot/port:ds0-group	Debugs the digital voice port you specify with the *slot/port:ds0-group* designation. *slot* 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. *port* 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.) *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 series with analog voice ports:

slot/port	Debugs the analog voice port you specify with the *slot/port* designation. *slot* 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. *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 series with digital voice ports:

slot:ds0-group	Debugs the digital voice port you specify with the *slot:ds0-group* designation. *slot* specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). *ds0-group* specifies a T1 or E1 logical voice port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

Defaults

Debug vtsp commands are not limited to a specific port.

Command History

Release	Modification
12.0(3)XG	This command was introduced on Cisco 2600 and 3600 series routers.
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco MC3810 series.

Usage Guidelines

Use this command to limit the debug output to a particular voice port. The debug output can be quite voluminous for a single channel. The entire vtsp debug output form a platform with 12 voice ports might create problems. Use this debug with any or all of the other debug modes.

Execution of **no debug vtsp all** will turn off all VTSP-level debugging. It is usually a good idea to turn off all debugging and then enter the debug commands you are interested in one by one. This will help to avoid confusion about which ports you are actually debugging.

Examples

The following example shows sample output from the **debug vtsp port 1/1/0** command:

```
router# debug vtsp port 1/1/0
*Mar  1 03:17:33.691: vtsp_tsp_call_setup_ind (sdb=0x613FD514, tdm_info=0x0,
   tsp_info=0x613FD438, calling_number= called_number= redirect_number=): peer_tag=1110
*Mar  1 03:17:33.691: vtsp_do_call_setup_ind
```

```
*Mar  1 03:17:33.691: dsp_close_voice_channel: [] packet_len=8 channel_id=1
   packet_id=75
*Mar  1 03:17:33.691: dsp_open_voice_channel: [] packet_len=12
   channel_id=1 packet_id=74 alaw_ulaw_select=0 transport_protocol=2
*Mar  1 03:17:33.695: dsp_set_playout_delay: [] packet_len=18
   channel_id=1 packet_id=76 mode=1 initial=60 min=4 max=200 fax_nom=300
*Mar  1 03:17:33.695: dsp_echo_canceller_control: [] packet_len=10 channel_id=1
   packet_id=66 flags=0x0
*Mar  1 03:17:33.695: dsp_set_gains: [] packet_len=12 channel_id=1 packet_id=91
   in_gain=0 out_gain=65506
*Mar  1 03:17:33.695: dsp_vad_enable: [] packet_len=10 channel_id=1 packet_id=78
   thresh=-38
*Mar  1 03:17:33.695: vtsp_process_event(): [, 0.S_SETUP_INDICATED, E_CC_PROCEEDING]
*Mar  1 03:17:33.699: vtsp_process_event(): [, 0.S_SETUP_INDICATED,
   E_CC_BRIDGE]act_bridge
*Mar  1 03:17:33.699: vtsp_ring_noan_timer_start: 1185370
*Mar  1 03:17:33.699: vtsp_process_event(): [, 0.S_SETUP_INDICATED,
   E_CC_CAPS_IND]act_caps_ind
*Mar  1 03:17:33.699: act_caps_ind: Encap 2, Vad 2, Codec 0x1000, CodecBytes 60,
              FaxRate 2, FaxBytes 30,
              Sub-channel 10, Bitmask 0x0 SignalType 2
*Mar  1 03:17:33.703: vtsp_process_event(): [, 0.S_SETUP_INDICATED,
   E_CC_CAPS_ACK]act_caps_ack
*Mar  1 03:17:33.703: dsp_idle_mode: [] packet_len=8 channel_id=1 packet_id=68
*Mar  1 03:17:33.703: vtsp_process_event(): [, 0.S_SETUP_INDICATED,
   E_CC_CONNECT]act_connect
*Mar  1 03:17:33.703: vtsp_ring_noan_timer_stop: 1185370
*Mar  1 03:17:33.911: vtsp_process_event(): [, 0.S_CONNECT, E_DSPRM_PEND_SUCCESS]
   act_pend_codec_success
*Mar  1 03:17:33.911: dsp_close_voice_channel: [] packet_len=8 channel_id=1
   packet_id=75
*Mar  1 03:17:33.911: dsp_open_voice_channel: [] packet_len=12 channel_id=1
   packet_id=74 alaw_ulaw_select=0 transport_protocol=2
*Mar  1 03:17:33.911: dsp_set_playout_delay: [] packet_len=18 channel_id=1 packet_id=76
   mode=1 initial=60 min=4 max=200 fax_nom=300
*Mar  1 03:17:33.911: dsp_echo_canceller_control: [] packet_len=10 channel_id=1
   packet_id=66 flags=0x0
*Mar  1 03:17:33.911: dsp_set_gains: [] packet_len=12 channel_id=1 packet_id=91
   in_gain=0 out_gain=65506
*Mar  1 03:17:33.911: dsp_vad_enable: [] packet_len=10 channel_id=1 packet_id=78
   thresh=-38
*Mar  1 03:17:33.911: dsp_encap_config: [] packet_len=24 channel_id=1 packet_id=
 92 TransportProtocol 3 SID_support=0 sequence_number=0 rotate_flag=0 header_bytes 0xA0
*Mar  1 03:17:33.915: dsp_voice_mode: [] packet_len=22 channel_id=1 packet_id=73
 coding_type=14 voice_field_size=60 VAD_flag=1 echo_length=128
 comfort_noise=1 fax_detect=1 digit_relay=0
```

Related Commands

Command	Description
debug vpm all	Enables all VPM debugging.
show debug	Shows which debug commands are enabled.

debug vtsp session

Use the **debug vtsp session** command to trace how the router interacts with the DSP based on the signaling indications from the signaling stack and requests from the application. Use the **no** form of this command to turn off the debug function.

debug vtsp session
no debug vtsp session

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp session is not enabled.

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco 2600, 3600 and MC3810 platforms.

Usage Guidelines

The **debug vtsp session** command traces how the router interacts with the DSP based on the signaling indications from the signaling stack and requests from the application. This debug command displays information about how each network indication and application request is handled, signaling indications, and DSP control messages.

This debug level shows the internal workings of the voice telephony call state machine.

Sample Display

The following example shows sample output from the **debug vtsp session** command, in which the call has been accepted and the system is checking for incoming dial-peer matches:

```
*Nov 30 00:46:19.535: vtsp_tsp_call_accept_check (sdb=0x60CD4C58,
calling_number=408 called_number=1): peer_tag=0
*Nov 30 00:46:19.535: vtsp_tsp_call_setup_ind (sdb=0x60CD4C58,
tdm_info=0x60B80044, tsp_info=0x60B09EB0, calling_number=408 called_number=1):
peer_tag=1
```

The following example shows sample output from the **debug vtsp session** command, in which a DSP has been allocated to handle the call and has indicated the call to the higher layer code:

```
*Nov 30 00:46:19.535: vtsp_do_call_setup_ind:
*Nov 30 00:46:19.535: dsp_open_voice_channel: [0:D:12] packet_len=12
channel_id=8737 packet_id=74 alaw_ulaw_select=0 transport_protocol=2
*Nov 30 00:46:19.535: dsp_set_playout_delay: [0:D:12] packet_len=18
channel_id=8737 packet_id=76 mode=1 initial=60 min=4 max=200 fax_nom=300
*Nov 30 00:46:19.535: dsp_echo_canceller_control: [0:D:12] packet_len=10
channel_id=8737 packet_id=66 flags=0x0
*Nov 30 00:46:19.539: dsp_set_gains: [0:D:12] packet_len=12 channel_id=8737
packet_id=91 in_gain=0 out_gain=0
*Nov 30 00:46:19.539: dsp_vad_enable: [0:D:12] packet_len=10 channel_id=8737
packet_id=78 thresh=-38
*Nov 30 00:46:19.559: vtsp_process_event: [0:D:12, 0.3, 13] act_setup_ind_ack
```

The following example shows sample output from the **debug vtsp session** command, in which the higher layer code has accepted the call, placed the DSP in DTMF mode, and collected digits:

```
*Nov 30 00:46:19.559: dsp_voice_mode: [0:D:12] packet_len=20 channel_id=8737
packet_id=73 coding_type=1 voice_field_size=160 VAD_flag=0 echo_length=64
comfort_noise=1 fax_detect=1
*Nov 30 00:46:19.559: dsp_dtmf_mode: [0:D:12] packet_len=10 channel_id=8737
packet_id=65 dtmf_or_mf=0
*Nov 30 00:46:19.559: dsp_cp_tone_on: [0:D:12] packet_len=30 channel_id=8737
packet_id=72 tone_id=3 n_freq=2 freq_of_first=350 freq_of_second=440
amp_of_first=4000 amp_of_second=4000 direction=1 on_time_first=65535
off_time_first=0 on_time_second=65535 off_time_second=0
*Nov 30 00:46:19.559: vtsp_timer: 278792
*Nov 30 00:46:22.059: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:22.059: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:22.059: vtsp_timer: 279042
*Nov 30 00:46:22.363: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:22.363: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:22.363: vtsp_timer: 279072
*Nov 30 00:46:22.639: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:22.639: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:22.639: vtsp_timer: 279100
*Nov 30 00:46:22.843: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:22.843: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:22.843: vtsp_timer: 279120
*Nov 30 00:46:23.663: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:23.663: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:23.663: vtsp_timer: 279202
```

The following example shows sample output from the **debug vtsp session** command, in which the call proceeded and DTMF was disabled:

```
*Nov 30 00:46:23.663: vtsp_process_event: [0:D:12, 0.4, 15] act_dcollect_proc
*Nov 30 00:46:23.663: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:23.663: dsp_idle_mode: [0:D:12] packet_len=8 channel_id=8737
packet_id=68
```

The following example shows sample output from the **debug vtsp session** command, in which the telephony call leg was conferenced with the packet network call leg, and the telephony call leg has performed capabilities exchange with the network-side call leg:

```
*Nov 30 00:46:23.699: vtsp_process_event: [0:D:12, 0.5, 17] act_bridge
*Nov 30 00:46:23.699: vtsp_process_event: [0:D:12, 0.5, 22] act_caps_ind
*Nov 30 00:46:23.699: vtsp_process_event: [0:D:12, 0.5, 23] act_caps_ack
Go into voice mode with codec indicated in caps exchange.
*Nov 30 00:46:23.699: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:23.699: dsp_idle_mode: [0:D:12] packet_len=8 channel_id=8737
packet_id=68
*Nov 30 00:46:23.699: dsp_voice_mode: [0:D:12] packet_len=20 channel_id=8737
packet_id=73 coding_type=6 voice_field_size=20 VAD_flag=1 echo_length=64
comfort_noise=1 fax_detect=1
```

The following example shows sample output from the **debug vtsp session** command in which the call has been connected at remote end:

```
*Nov 30 00:46:23.779: vtsp_process_event: [0:D:12, 0.5, 10] act_connect
```

The following example shows sample output from the **debug vtsp session** command in which disconnect was indicated and passed to upper layer:

```
*Nov 30 00:46:30.267: vtsp_process_event: [0:D:12, 0.11, 5] act_generate_disc
```

The following example shows sample output from the **debug vtsp session** command, in which the conference was torn down and the disconnect handshake was completed:

```
*Nov 30 00:46:30.267: vtsp_process_event: [0:D:12, 0.11, 18] act_bdrop
*Nov 30 00:46:30.267: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:30.267: vtsp_process_event: [0:D:12, 0.11, 20] act_disconnect
*Nov 30 00:46:30.267: dsp_get_error_stat: [0:D:12] packet_len=10 channel_id=0
packet_id=6 reset_flag=1
*Nov 30 00:46:30.267: vtsp_timer: 279862
```

The following example shows sample output from the **debug vtsp session** command, in which the final DSP statistics were retrieved:

```
*Nov 30 00:46:30.275: vtsp_process_event: [0:D:12, 0.17, 30] act_get_error
*Nov 30 00:46:30.275: 0:D:12: rx_dropped=0 tx_dropped=0 rx_control=353
tx_control=338 tx_control_dropped=0 dsp_mode_channel_1=2 dsp_mode_channel_2=0
c[0]=71 c[1]=71 c[2]=71 c[3]=71 c[4]=68 c[5]=71 c[6]=68 c[7]=73 c[8]=83 c[9]=84
c[10]=87 c[11]=83 c[12]=84 c[13]=87 c[14]=71 c[15]=6
*Nov 30 00:46:30.275: dsp_get_levels: [0:D:12] packet_len=8 channel_id=8737
packet_id=89
*Nov 30 00:46:30.279: vtsp_process_event: [0:D:12, 0.17, 34] act_get_levels
*Nov 30 00:46:30.279: dsp_get_tx_stats: [0:D:12] packet_len=10 channel_id=8737
packet_id=86 reset_flag=1
*Nov 30 00:46:30.287: vtsp_process_event: [0:D:12, 0.17, 31] act_stats_complete
*Nov 30 00:46:30.287: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:30.287: dsp_idle_mode: [0:D:12] packet_len=8 channel_id=8737
packet_id=68
*Nov 30 00:46:30.287: vtsp_timer: 279864
```

The following example shows sample output from the **debug vtsp session** command, in which the DSP channel was closed and released:

```
*Nov 30 00:46:30.287: vtsp_process_event: [0:D:12, 0.18, 6] act_wrelease_release
*Nov 30 00:46:30.287: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:30.287: dsp_idle_mode: [0:D:12] packet_len=8 channel_id=8737
packet_id=68
*Nov 30 00:46:30.287: dsp_close_voice_channel: [0:D:12] packet_len=8
channel_id=8737 packet_id=75
*Nov 30 00:46:30.287: vtsp_process_event: [0:D:12, 0.16, 42] act_terminate
```

Related Commands

Command	Description
debug vpm all	Enables all VPM debugging.
debug vtsp port	Limits vtsp debug output to a specific voice port.
show debug	Shows which debug commands are enabled.

debug vtsp stats

Use the **debug vtsp stats** command to debug periodic statistical-information-request messages sent and received from the DSP during a call. Use the **no** form of this command to turn off the debug function.

debug vtsp stats
no debug vtsp stats

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp stats is not enabled.

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco 2600, 3600 and MC3810 platforms.

Usage Guidelines

The **debug vtsp stats** command generates a collection of DSP statistics for generating RTCP packets and a collection of other statistical information.

Sample Display

The following example shows sample **debug vtsp stats** output:

```
*Nov 30 00:53:26.499: vtsp_process_event: [0:D:14, 0.11, 19] act_packet_stats
*Nov 30 00:53:26.499: dsp_get_voice_playout_delay_stats: [0:D:14] packet_len=10
channel_id=8753 packet_id=83 reset_flag=0
*Nov 30 00:53:26.499: dsp_get_voice_playout_error_stats: [0:D:14] packet_len=10
channel_id=8753 packet_id=84 reset_flag=0
*Nov 30 00:53:26.499: dsp_get_rx_stats: [0:D:14] packet_len=10 channel_id=8753
packet_id=87 reset_flag=0
*Nov 30 00:53:26.503: vtsp_process_dsp_message: MSG_TX_GET_VOICE_PLAYOUT_DELAY:
clock_offset=-1664482334 curr_rx_delay_estimate=69 low_water_mark_rx_delay=69
high_water_mark_rx_delay=70
*Nov 30 00:53:26.503: vtsp_process_event: [0:D:14, 0.11, 28]
act_packet_stats_res
*Nov 30 00:53:26.503: vtsp_process_dsp_message: MSG_TX_GET_VOICE_PLAYOUT_ERROR:
predective_concelement_duration=0 interpolative_concelement_duration=0
silence_concelement_duration=0 retroactive_mem_update=0
buf_overflow_discard_duration=10 num_talkspurt_detection_errors=0
*Nov 30 00:53:26.503: vtsp_process_event: [0:D:14, 0.11, 29]
act_packet_stats_res
*Nov 30 00:53:26.503: vtsp_process_dsp_message: MSG_TX_GET_RX_STAT:
num_rx_pkts=152 num_early_pkts=-2074277660 num_late_pkts=327892
num_signalling_pkts=0 num_comfort_noise_pkts=0 receive_durtation=3130
voice_receive_duration=2970 fax_receive_duration=0 num_pack_ooseq=0
num_bad_header=0
*Nov 30 00:53:26.503: vtsp_process_event: [0:D:14, 0.11, 32]
act_packet_stats_res
```

Related Commands

Command	Description
debug vpm all	Enables all VPM debugging.
debug vtsp port	Limits vtsp debug output to a specific voice port.
show debug	Shows which debug commands are enabled.

debug vtsp vofr subframe

To display the first 10 bytes (including header) of selected VoFR subframes for the interface, use the **debug vtsp vofr subframe** command. Use the **no** form of the command to turn off the debug function.

debug vtsp vofr subframe *payload* **[from-dsp] [to-dsp]**
no debug vtsp vofr subframe

Syntax Description

payload	Number used to selectively display subframes of a specific payload. The payload types are: 0: Primary Payload - WARNING! This option may cause network instability 1: Annex-A 2: Annex-B 3: Annex-D 4: All other payloads 5: All payloads - WARNING! This option may cause network instability
from-dsp	Displays only the subframes received from the DSP.
to-dsp	Displays only the subframes going to the DSP.

Defaults

Debugging for vtsp vofr subframe is not enabled.

Command History

Release	Modification
12.0(3)XG, 12.0(4)T	This command was introduced on the Cisco 2600 and 3600 platforms.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

Each debug output displays the first 10 bytes of the FRF.11 subframe, including header bytes. The **from-dsp** and **to-dsp** options can be used to limit the debugs to a single direction. If not specified, debugs are displayed for subframes when they are received from the DSP and before they are sent to the DSP.

Use extreme caution in selecting payload options 0 and 6. These options may cause network instability.

Examples

The following example shows sample output from the **debug vtsp vofr subframe** command:

```
router# debug vtsp vofr subframe 2
vtsp VoFR subframe debugging is enabled for payload 2 to and from DSP 3620_vofr#
*Mar  6 18:21:17.413:VoFR frame received from Network (24 bytes):9E 02 19 AA AA AA AA
      AA AA AA
*Mar  6 18:21:17.449:VoFR frame received from DSP (18 bytes):9E 02 19 AA AA AA AA AA AA
      AA
*Mar  6 18:21:23.969:VoFR frame received from Network (24 bytes):9E 02 19 AA AA AA AA
      AA AA AA
*Mar  6 18:21:24.005:VoFR frame received from DSP (18 bytes):9E 02 19 AA AA AA AA AA AA
      AA
```

Related Commands

Command	Description
debug vpm all	Enables all VPM debugging.
debug vtsp port	Limits vtsp debug output to a specific voice port.
show debug	Shows which debug commands are enabled.