Voice over IP for the Cisco AS5800

The Voice over IP for the Cisco AS5800 feature adds Voice over IP carrier-class gateway functionality to the Cisco AS5800 platform. This document contains the following sections:

Feature Overview

Voice over IP (VoIP) enables a Cisco AS5800 universal access server to provide voice and fax traffic, such as telephone calls and faxes, over an IP network. There are basically two different environments in which VoIP can be deployed: enterprise and service provider. Different strategies have been developed for deploying VoIP in both of these environments. The Cisco AS5800 universal access server can be configured for deployment in either an enterprise or a service provider environment but, because of the extensive capabilities of the Cisco AS5800 universal access server, it is more likely that it will function as a carrier class gateway in a service provider environment. This document, then, describes how to configure the Cisco AS5800 universal access server to act as a carrier class gateway in your VoIP network. To configure the Cisco AS5800 universal access server to perform in an enterprise environment, refer to the Cisco IOS Release 12.0(3)T *Voice over IP for the Cisco AS5300* feature module. The configuration steps for both the Cisco AS5300 access server and the Cisco AS5800 universal access server for an enterprise environment are identical.

Voice over IP in either the service provider or enterprise environment is primarily a software feature; however, to use this feature on the Cisco AS5800, you must install a VoIP feature card (VFC). The VFC uses the Cisco AS5800's T1/E1 and T3 Public Switched Telephone Network (PSTN) interfaces and local-area network (LAN) or wide-area network (WAN) routing capabilities to provide up to a 192 ports or channels (per VFC card) for VoIP packetized voice traffic.

Benefits

Two-Stage-Dial Toll Bypass

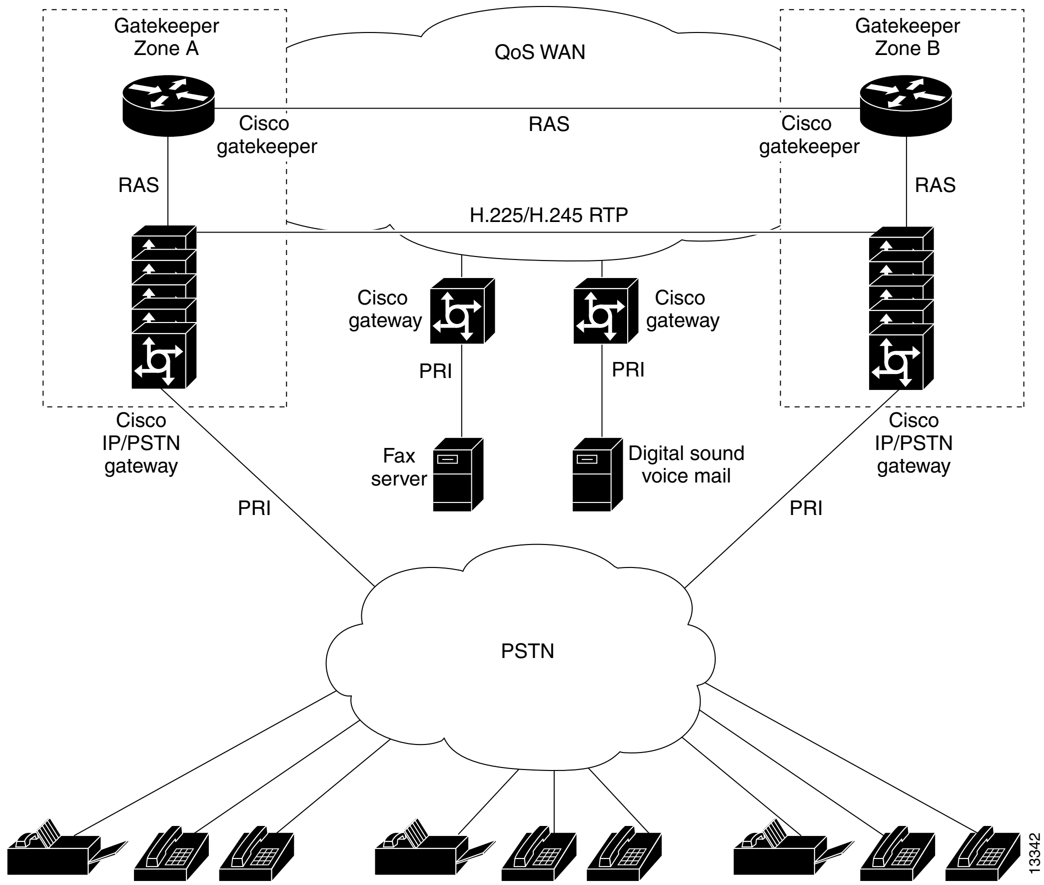
With Voice over IP on the Cisco AS5800, you can leverage your network's WAN infrastructure to offer long distance toll bypass services. Toll bypass occurs in two stages. For example, customers can be assigned an account number and a Personal Identification Number (PIN). When a user dials a local number or a 1-800-Internet Telephone Service Provider (ITSP) number, she connects to the local VoIP point of presence. She is then prompted by the Interactive Voice Response (IVR) to input her account and PIN numbers. Following authentication, a second dial tone allows her to enter an E.164 destination telephone number.

The local gatekeeper maps the E.164 destination telephone number to an IP address of a remote-zone gatekeeper, which then selects a gateway to terminate the call. The gateway encodes the call, encapsulates it in Real Time Protocol (RTP) packets and routes it over the WAN to the remote gateway. The remote gateway decodes the call and delivers it to the receiver.

For information about configuring IVR, refer to the Cisco IOS Release 12.0(7)T *Configuring Interactive Voice Response for Cisco Access Platforms* feature module.

Figure 1 illustrates this benefit.

Figure 1 Two-Stage Dial Toll Bypass

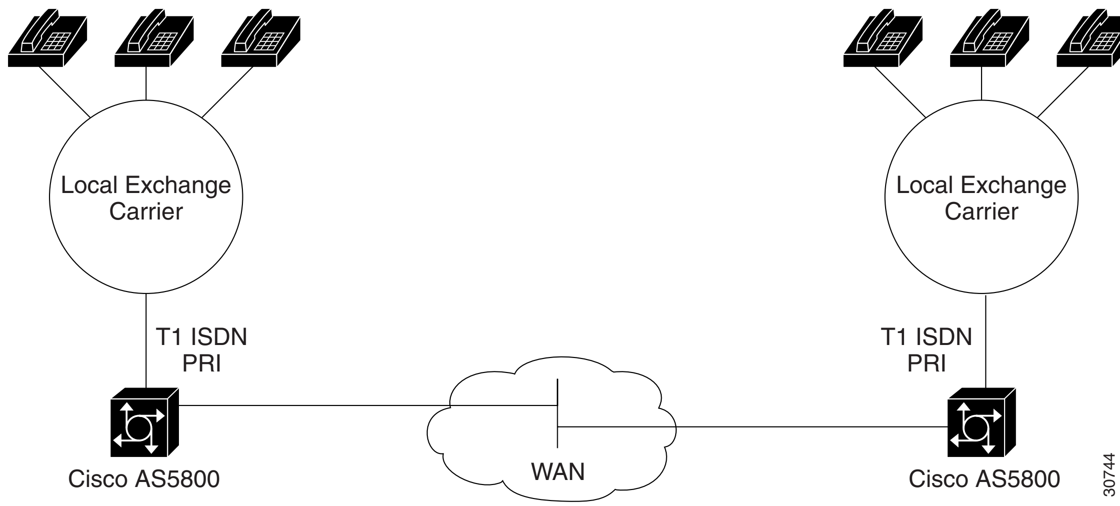


PSTN Voice-Traffic and Fax-Traffic Off load

Carriers can leverage their WAN infrastructure to off load voice and fax traffic from their congested PSTN networks by using the Cisco AS5800 as a carrier class voice gateway. In this application, PSTN traffic designated to be off-loaded is forwarded to a tandem switch connected to the Cisco AS5800 gateway. The AS5800 gateway then encapsulates the off-loaded PSTN traffic into RTP streams and routes it across the WAN.

The signaling interface between the PSTN and the Cisco AS5800 can be either Common Channel Signaling (CCS), with SS7 terminated by the VCO-4K service point or Channel Associated Signaling (CAS), configured in Direct Inward Dial (DID) mode. Figure 2 illustrates this application.

Figure 2 VoIP Used as a PSTN Gateway to Off load Voice Traffic and Fax Traffic



Universally Accessible Voice-Mail and Fax-Mail Services

VoIP on the Cisco AS5800 can be used to leverage the technology prefixes feature. Gateways (with voice/fax feature cards) that are connected to the voice-mail and fax-mail servers can be identified by gatekeepers based on a prefix prepended to an E.164 telephone number.

Additional Benefits

VoIP on the Cisco AS5800 can be used to provide the following additional benefits:

- Remote PBX presence over WANs
- POTS-Internet telephony gateways

Restrictions

To run Voice over IP on the Cisco AS5800, the AS5800 must have a version of the Cisco IOS software installed that supports DSDWare 3.1.7 (for example, Cisco IOS Release 12.0(4)XL or Cisco IOS Release 12.0(7)T).

Related Features and Technologies

- Cisco IOS Release 12.0(3)T *Voice over IP for the Cisco AS5300* feature module
- Cisco IOS Release 12.0(3)T *Service Provider Features for Voice over IP* feature module
- Cisco IOS Release 12.0(5)T *IP RTP Priority* feature module
- Cisco IOS Release 12.0(7)T *Configuring Interactive Voice Response for Cisco Access Platforms* feature module

Related Documents

- *Voice, Video, and Home Applications Configuration Guide*, Cisco IOS Release 12.0

- *Voice, Video, and Home Applications Command Reference*, Cisco IOS Release 12.0
- *Quality of Service Configuration Guide*, Cisco IOS Release 12.0
- *Quality of Service Command Reference*, Cisco IOS Release 12.0
- *Voice over IP for the Cisco AS5800 Software Configuration Guide*, Cisco IOS Release 12.0(4)XL.

Supported Platforms

- Cisco AS5800 universal access servers
- Cisco AS5300 access servers
- Cisco 2600 series routers
- Cisco 3600 series routers

Supported Standards, MIBs, and RFCs

Standards

None

MIBs

- IF-MIB
- ENTITY-MIB.my
- CISCO-ENTITY-VENDORTYPE-OID-MIB.my
- DIAL-CONTROL-MIB.my
- CISCO-DIAL-CONTROL-MIB.my
- CISCO-VOICE-DIAL-CONTROL-MIB.my
- CISCO-VOICE-IF-MIB.my
- CISCO-DSP-MGMT-MIB.my
- CISCO-MMAIL-DIAL-CONTROL-MIB.my
- CISCO-CAS-IF-MIB.my

For descriptions of supported MIBs and how to use MIBs, see the Cisco MIB web site on CCO at http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml.

RFCs

None

Prerequisites

Before you can configure your Cisco AS5800 to use Voice over IP, you must first:

- Install a version of the Cisco IOS software that supports DSPWare 3.1.7 specific to the Cisco AS5800 (for example, Cisco IOS Release 12.0(4)XL or Cisco IOS Release 12.0(7)T).
- Establish a working IP network. For more information about configuring IP, refer to the "IP Overview," "Configuring IP Addressing," and "Configuring IP Services" chapters in the Cisco IOS 12.0 *Network Protocols Configuration Guide, Part 1*.
- Complete basic configuration for the AS5800. This includes, as a minimum, the following tasks:
  - Configure a host name and password for the AS5800
  - Configure the Fast Ethernet interface of your AS5800 so that it can be recognized as a device on the Ethernet LAN
  - Configure the AS5800 interfaces for ISDN PRI lines
  - Configure the ISDN D channels for each ISDN PRI line
  - Configure the AS5800 interfaces for T1 CAS lines
  - Configure the ISDN D channels for each T1 CAS PRI line

  For more information about any of the these configuration tasks, refer to the *Cisco AS5800 Universal Access Server Software Installation and Configuration Guide*, which shipped with your Cisco AS5800 and is available on the document CD-ROM.
- Install the VFC into the appropriate slot of your Cisco AS5800 universal access server. Each VFC can hold up to 16 digital signal processor modules (DSPMs), enabling processing for up to 192 voice channels. For more information about the physical characteristics of the VFCs or DSPMs, or how to install them, refer to *Installing Voice over IP Feature Cards in Cisco AS5800 Universal Access Servers* document that shipped with your VFC and is available online.
- Complete your company's dial plan.
- Establish a working telephony network based on your company's dial plan.

- Integrate your dial plan and telephony network into your existing IP network topology. Merging your IP and telephony networks depends on your particular IP and telephony network topology. In general, we recommend the following suggestions:
  - Use canonical numbers wherever possible. It is important that you avoid situations where numbering systems are significantly different on different routers or access servers in your network.
  - Make routing and dialing transparent to the user. For example, avoid secondary dial tones from secondary switches, where possible.
  - Contact your PBX vendor for instructions about how to reconfigure the appropriate PBX interfaces.
- Configure another device in your network (preferably a Cisco 2600 or Cisco 3600 series router) to act as a gatekeeper. The Service Provider implementation of Voice over IP is configured using both gatekeepers and gateways. Because of the extensive capabilities of the Cisco AS5800 universal access server, it is likely that it will function as a carrier class gateway in a Service Provider environment. Unless it has a gatekeeper to interact with, it will periodically query all devices in the network, searching for a gatekeeper. For more information about configuring gatekeepers, refer to the Cisco IOS Release 12.0(3)T *Service Provider Features for Voice over IP* feature module.

Configuration Tasks

After you have analyzed your dial plan and decided how to integrate it into your existing IP network, you are ready to configure your network devices to support Voice over IP. The actual configuration procedure depends entirely on the topology of your voice network, but, in general, you need to complete the following tasks:

- Configuring IP Networks for Real-Time Voice Traffic
- Configuring Voice Ports
- Configuring Dial Peers
- Configuring the Cisco AS5800 as an H.323 Gateway
- Configuring the Cisco AS5800 for Interactive Voice Response

Configuring IP Networks for Real-Time Voice Traffic

You need to have a well-engineered network end-to-end when running delay-sensitive applications such as VoIP. Fine-tuning your network to adequately support VoIP involves a series of protocols and features geared toward Quality of Service (QoS). It is beyond the scope of this document to explain the specific details relating to wide-scale QoS deployment. Cisco IOS software provides many tools for enabling QoS on your backbone, such as Random Early Detection (RED), Weighted Random Early Detection (WRED), Fancy Queuing (meaning custom, priority, or weighted fair queuing), and IP Precedence. To configure your IP network for real-time voice traffic, you need to take into consideration the entire scope of your network, then select the appropriate QoS tool or tools. In addition, you must use the Cisco IOS **ip cef** command to ensure that Cisco Express Forwarding (CEF) is enabled.

QoS must be configured throughout your network—not just on the Cisco AS5800 devices running VoIP—to improve voice network performance. Not all QoS techniques are appropriate for all network routers. Edge routers and backbone routers in your network do not necessarily perform the same operations; the QoS tasks they perform might also differ. To configure your IP network for real-time voice traffic, you need to consider the functions of both edge and backbone routers in your network, then select the appropriate QoS tool or tools.

In general, edge routers perform the following QoS functions:

- Packet classification
- Admission control
- Bandwidth management
- Queuing

In general, backbone routers perform the following QoS functions:

- High-speed switching and transport
- Congestion management
- Queue management

Scalable QoS solutions require cooperative edge and backbone functions.

Configuring Custom Queuing and IP RTP Reserve

Although not required, you can use the custom queuing QoS tool to fine-tune your network for real-time voice traffic. Real-time voice traffic is carried on UDP ports ranging from 16384 to 32767. Custom Queuing and other methods for identifying high priority streams should be configured for these port ranges. For more information about custom queuing, refer to the "Congestion Management" chapter in the Cisco IOS Release 12.0 *Quality of Service Configuration Guide*. For more information about configuring IP RTP Priority, refer to the Cisco IOS Release 12.0(5)T *IP RTP Priority* feature module.

Configuring Voice Ports

When an ISDN interface on the Cisco AS5800 is carrying voice data, it is referred to as a voice port.

Note A voice port was created automatically when you installed the VFC in the Cisco AS5800 and configured an ISDN PRI group. Configuring an ISDN PRI group is part of the basic Cisco AS5800 configuration procedure. For more information, refer to the *Cisco AS5800 Universal Access Server Software Installation Configuration Guide*.

Signaling in Voice over IP for the AS5800 is handled by ISDN PRI group configuration. After ISDN PRI is configured for both B and D channels for both ISDN PRI lines, you need to issue the **isdn incoming-voice** command on the serial interface (acting as the D channel) to ensure a dial tone.

Under most circumstances, the default voice-port command values are adequate to configure voice ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, you might need specific voice-port values configured, depending on the specifications of the devices in your telephony network. For more information on specific voice-port configuration commands, refer to either the Cisco IOS Release 12.0(3)T *Voice over IP for the Cisco AS5300* feature module or the Cisco IOS Release 12.0 *Voice, Video, and Home Applications Command Reference*.

To configure basic ISDN parameters for Voice over IP on the Cisco AS5800, perform the following steps:

Step	Command	Purpose
1	`Router(config)#` **`isdn switch-type`** *`switch-type`*	Defines the telephone company's switch type.
2	`Router(config)#` **`controller T1 1/0/0`** or `Router(config)#` **`controller T1 1/0/0:1`**	Enables the T1 0 controller on the T1 card and enters controller configuration mode, or Enables the T1 1 controller on the T3 card and enters controller configuration mode.
3	`Router(config)#` **`framing esf`**	Defines the framing characteristics.
4	`Router(config)#` **`linecode`** *`value`*	Sets the line code type to match that of your telephone company service provider.
5	`Router(config)#` **`pri-group timeslots`** *`range`*	Configures ISDN PRI.
6	`Router(config)#` **`controller T1 1/0/1`** or `Router(config)#` **`controller T1 1/0/0:2`**	Enables the T1 1 on the T1 card controller and enters controller configuration mode, or Enables the T1 2 controller on the T3 card and enters controller configuration mode.
7	`Router(config)#` **`framing esf`**	Defines the framing characteristics.
8	`Router(config)#` **`linecode`** *`value`*	Sets the line code type to match that of your telephone company service provider.
9	`Router(config)#` **`pri-group timeslots`** *`range`*	Configures ISDN PRI.
10	`Router(config)#` **`interface Serial1/0/0:23`** or `Router(config)#` **`interface Serial1/0/0:1:23`**	Configures the channel for the first ISDN PRI line on the T1 card. (The ISDN serial interface is the D channel.) or Configures the channel for the first ISDN PRI line on the T3 card.
11	`Router(config)#` **`isdn incoming-voice modem`**	Enables incoming ISDN voice calls. This command has two possible keywords: **data** and **modem**. You must use the **modem** keyword to enable voice calls. The **modem** keyword represents bearer capabilities of speech.
12	`Router(config)#` **`interface Serial1/0/1:23`** or `Router(config)#` **`interface Serial1/0/0:2:23`**	Configures the channel for the second ISDN PRI line.or Configures the channel for the second ISDN PRI line on the T3 card.
13	`Router(config)#` **`isdn incoming-voice modem`**	Enables incoming ISDN voice calls. This command has two possible keywords: **data** and **modem**. You must use the **modem** keyword to enable voice calls. The **modem** keyword represents bearer capabilities of speech.

As mentioned, under most circumstances, the default voice-port command values are adequate to configure voice ports to transport voice data over your existing IP network. If you need to configure specific voice port parameters, perform the following steps beginning in privileged EXEC mode:

Step	Command	Purpose
1	`Router#` **`configure terminal`**	Enters global configuration mode.
2	`Router(config)#` **`voice-port`** `{`*`shelf`*`/`*`slot`*`/`*`port`***`:D`**`}|` `{`*`shelf`*`/`*`slot`*`/`*`parent`***`:`***`port`***`:D`**`}`	Identifies the voice port you want to configure and enters voice-port configuration mode.

Step	Command	Purpose
3	`Router(config-voiceport)#` **cptone** *country*	Selects the appropriate voice call progress tone for this interface. The default for this command is **us**. For a list of supported countries, refer to the *Multiservice Applications Command Reference*.
4	`Router(config-voiceport)#` **compand-type** {**a-law**|**u-law**}	Selects a companding type for this voice port.
5	`Router(config-voiceport)#` **connection** {**plar** *string*|**trunk** *string*}	(Optional) Specifies either the trunk connection or the private line auto ringdown (PLAR) connection. The *string* value specifies the destination telephone number.
6	`Router(config-voiceport)#` **music-threshold** *number*	(Optional) Specifies the threshold (in decibels) for on-hold music. Valid entries are from –70 to –30.
7	`Router(config-voiceport)#` **description** *string*	(Optional) Attaches descriptive text about this voice port connection.

Fine-Tuning ISDN Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain, and output attenuation for voice ports. Collectively, these commands are referred to as voice-port tuning commands.

Note In most cases, the default values for voice-port tuning commands will be sufficient.

To fine-tune ISDN voice ports, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	`Router#` **configure terminal**	Enters global configuration mode.
2	`Router(config)#` **voice-port** {*shelf*/*slot*/*port*:**D**} | {*shelf*/*slot*/*parent*:*port*:**D**}	Identifies the voice port you want to configure and enter voice-port configuration mode.
3	`Router(config-voiceport)#` **input gain** *value*	Specifies (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from –6 to 14.
4	`Router(config-voiceport)#` **output attenuation** *value*	Specifies (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14.
5	`Router(config-voiceport)#` **echo-canel enable**	Enables echo-cancellation of voice that is sent out the interface and received back on the same interface.
6	`Router(config-voiceport)#` **echo-canel coverage** *value*	Adjusts the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32.
7	`Router(config-voiceport)#` **non-linear**	Enables non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.)
8	`Router(config-voiceport)#` **playout-delay** {**maximum** *milliseconds*|**nominal** *milliseconds*}	Specifies the amount of time in milliseconds configured for the playout delay buffer.
9	`Router(config-voiceport)#` **timeouts initial** *seconds*	Specifies the number of seconds the system will wait for the caller to input the first digit of the dialed digits. Valid entries for this command are from 0 to 120.

Step	Command	Purpose
10	`Router(config-voiceport)#` **`timeouts interdigits`** *`seconds`*	Specifies the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit. Valid entries for this command are from 0 to 120.
11	`Router(config-voiceport)#` **`timeouts ringing`** `{`*`seconds`*`|`**`infinity`**`}`	Specifies the number of seconds the system will continue to ring the destination if there is no answer.
12	`Router(config-voiceport)#` **`timeouts wait-release`** `{`*`seconds`*`|`**`infinity`**`}`	Specifies the wait release timeout duration in seconds.
13	`Router(config-voiceport)#` **`translate`** `{`**`called`** *`number`*`|`**`calling`** *`number`*`}`	Defines translation rules pertaining to either the called or calling numbers.

For more information on specific voice-port configuration commands or additional voice-port commands, refer to either the Cisco IOS Release 12.0(3)T *Voice over IP for the Cisco AS5300* feature module or the Cisco IOS Release 12.0 *Voice, Video, and Home Applications Command Reference*..

Verifying Voice Port Configuration

- Use the **show voice port** command to verify that the data configured is correct.
- If you have not configured your device to support direct inward dial, dial in to the router and see if you have dial tone.
- Enter DTMF digit. If the dial tone stops, you have two-way voice connectivity with the router.

Troubleshooting Tips

If you are having trouble connecting a call, and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the “Configuring IP” chapter in the Cisco IOS 12.0 *Network Protocols Configuration Guide, Part 1*.
- Check to see that the VFC has been correctly installed.
- Use the **show dial-shelf** command to see if the VFC is operational.
- Use the **show vrm vdevices summary** command to verify that you have voice devices available.
- Use the **show isdn status** command to view layer status information. If you receive a status message stating that Layer 1 is deactivated, make sure the cable connection is not loose or disconnected. (This status message indicates a problem at the physical layer.)
- With T1 lines, check to see if your u-law setting is correct. With E1 lines, check to see if your a-law setting is correct. Use the **cptone** command to configure both a-law or u-law values. For more information about the **cptone** command, refer to the Cisco IOS Release 12.0(3)T *Voice over IP for the Cisco AS5300* feature module.
- If dialing cannot occur, use the **debug isdn q931** command to check the ISDN configuration.

Configuring Dial Peers

The key point to understanding how VoIP functions is to understand dial peers. Each dial peer defines the characteristics associated with a call leg, as shown in Figure 3 and Figure 4. A call leg is a discrete segment of a call connection that lies between two points in the connection. All of the call legs for a particular connection have the same connection ID.

There are two different kinds of dial peers:

- POTS—Dial peer describing the characteristics of a traditional telephony network connection. POTS peers point to a particular voice port on a voice network device.
- VoIP—Dial peer describing the characteristics of a packet network connection. VoIP peers point to specific VoIP devices.

An end-to-end call comprises four call legs, two from the perspective of the source access server as shown in Figure 3, and two from the perspective of the destination access server as shown in Figure 4. A dial peer is associated with each call leg. Dial peers are used to apply attributes to call legs and to identify call origin and destination. Attributes applied to a call leg include QoS, codec, VAD, and fax rate.

Figure 3 Dial Peer Call Legs from the Perspective of the Source Router

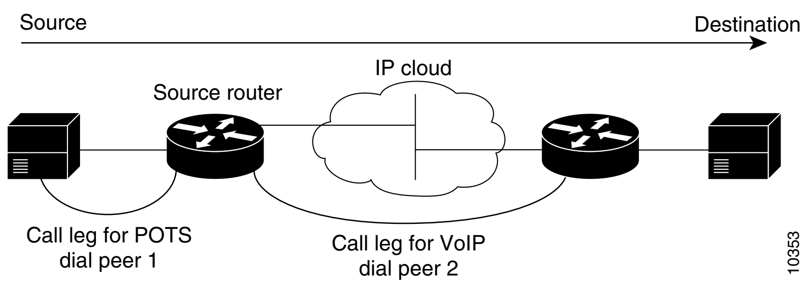
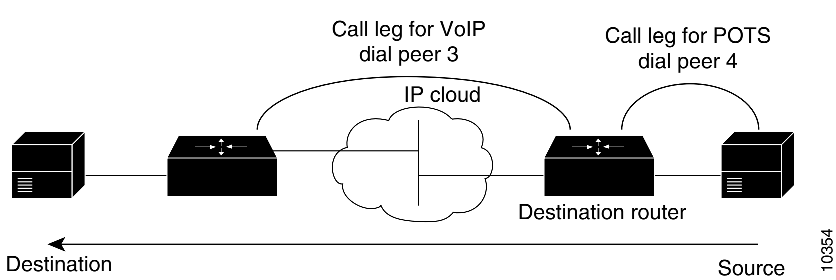


Figure 4 Dial Peer Call Legs from the Perspective of the Destination Router



Inbound versus Outbound Dial Peers

Dial peers are used for both inbound and outbound call legs. It is important to remember that these terms are defined from the *access server's* perspective. An inbound call leg originates *outside* the access server. An outbound call leg originates *from* the access server.

For inbound call legs, a dial peer might be associated to the calling number or the port designation. Outbound call legs always have a dial peer associated with them. The destination pattern is used to identify the outbound dial peer. The call is associated with the outbound dial peer at setup time.

POTS peers associate a telephone number with a particular voice port so that incoming calls for that telephone number can be received and outgoing calls can be placed. VoIP peers point to specific devices (by associating destination telephone numbers with a specific IP address) so that incoming calls can be received and outgoing calls can be placed. Both POTS and VoIP peers are needed to establish VoIP connections.

Configuring POTS Peers

POTS peers enable incoming calls to be received by a particular telephony device. To configure a POTS peer, you need to uniquely identify the peer (by assigning it a unique tag number), define its telephone numbers, and associate it with a voice port through which calls will be established. Under most circumstances, the default values for the remaining dial peer configuration commands will be sufficient to establish connections.

To configure a POTS dial peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	`Router(config)#` **`dial-peer voice`** *`number`* **`pots`**	Enters the dial peer configuration mode to configure a POTS peer. The *number* value of the **dial-peer voice pots** command is a tag that uniquely identifies the dial peer.
2	`Router(config-dial-peer)#` **`destination-pattern`** *`string`*	Defines the telephone number associated with this POTS dial peer.
3	`Router(config-dial-peer)#` **`port`** *`shelf/slot/port`***`:D`**	Associates this POTS dial peer with a specific logical dial interface.
4	`Router(config-dial-peer)#` **`prefix`** *`string`*	(Optional) Specifies the prefix for this POTS dial peer. The **prefix** *string* value is sent to the telephony interface first, before the telephone number (destination pattern) associated with this dial peer is sent.

For additional POTS dial-peer configuration commands, refer to the “Voice-Related Commands” section of the Cisco IOS Release 12.0 *Voice, Video, and Home Applications Command Reference*, the Cisco IOS Release 12.0(3)T *Voice over IP for the Cisco AS5300* feature module, and the Cisco IOS Release 12.0(3)T *Service Provider Features for Voice over IP* feature module.

Outbound Dialing on POTS Peers

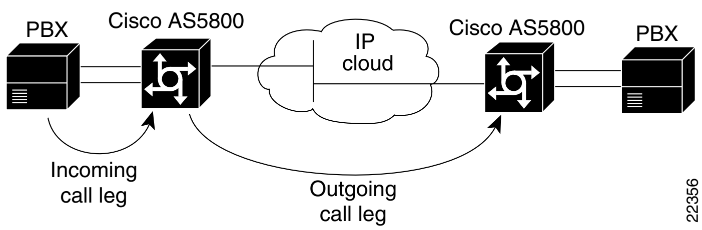
When a router receives a voice call, it selects an outbound dial peer by comparing the called number (the full E.164 telephone number) in the call information with the number configured as the destination pattern for the POTS peer. The router then strips out the left-justified numbers corresponding to the destination pattern matching the called number. If you have configured a prefix, the prefix will be put in front of the remaining numbers, creating a dial string, which the router will then dial. If all numbers in the destination pattern are stripped-out, the user will receive (depending on the attached equipment) a dial tone.

For example, suppose there is a voice call whose E.164 called number is 1 310 767-2222. If you configure a destination-pattern of “1310767” and a prefix of “9,” the router will strip out “1310767” from the E.164 telephone number, leaving the extension number of “2222.” It will then append the prefix, “9,” to the front of the remaining numbers, so that the actual numbers dialed is “9, 2222.” The comma in this example means that the router will pause for one second between dialing the “9” and the “2” to allow for a secondary dial tone.

Direct Inward Dial for POTS Peers

Direct inward dial (DID) is used to determine how the called number is treated for incoming POTS call legs. As shown in Figure 5, incoming means from the perspective of the router. In this case, it is the call leg coming into the access server to be forwarded through to the appropriate destination pattern.

Figure 5 Incoming and Outgoing POTS Call Legs



Unless otherwise configured, when a call arrives on the access server, the server presents a dial tone to the caller and collects digits until it can identify the destination dial peer. After the dial peer is identified, the call is forwarded through the next call leg to the destination.

There are cases where it might be necessary for the server to use the called-number (DNIS) to find a dial peer for the outgoing call leg—for example, if the switch connecting the call to the server has already collected the digits. DID enables the server to match the called-number with a dial peer and then directly place the outbound call. With DID, the server does not present a dial tone to the caller and does not collect digits; it forwards the call directly to the configured destination.

To use DID and incoming called-number, a dial peer must be associated with the incoming call leg. Before doing this, it helps if you understand the logic behind the algorithm used to associate the incoming call leg with the dial peer.

The algorithm used to associate incoming call legs with dial peers uses three inputs (which are derived from signaling and interface information associated with the call) and four defined dial peer elements. The three signaling inputs are:

- Called-number (DNIS)—Set of numbers representing the destination, which is derived from the ISDN setup message or CAS DNIS.
- Calling-number (ANI)—Set of numbers representing the origin, which is derived from the ISDN setup message or CAS DNIS.
- Voice port—The voice port carrying the call.

The four defined dial peer elements are:

- Destination pattern—A pattern representing the phone numbers to which the peer can connect.
- Answer address—A pattern representing the phone numbers from which the peer can connect.
- Incoming called-number—A pattern representing the phone numbers that associate an incoming call leg to a peer based on the called-number or DNIS.
- Port—The port through which calls to this peer are placed.

Using the elements, the algorithm is as follows:

```
For all peers where call type (VoIP versus POTS) match dial peer type:
if the type is matched, associate the called number with the incoming called-number
 else if the type is matched, associate calling-number with answer-address
 else if the type is matched, associate calling-number with destination-pattern
 else if the type is matched, associate voice port to port
```

This algorithm shows that if a value is not configured for answer-address, the origin address is used because, in most cases, the origin address and answer-address are the same.

To configure a POTS dial peer for direct inward dial, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	`Router(config)#` **`dial-peer voice`** *`number`* **`pots`**	Enters the dial peer configuration mode to configure a POTS peer.
2	`Router(config-dial-peer)#` **`direct-inward-dial`**	Specifies direct inward dial for this POTS peer.

Note Direct inward dial is configured for the calling POTS dial peer.

Distinguishing Voice and Modem Calls on the Cisco AS5800

When the Cisco AS5800 is handling both modem and voice calls, it needs to be able to identify the service type of the call—that is, whether or not the incoming call to the server is a modem or a voice call. When the access server handles only modem calls, the service type identification is handled through modem pools. Modem pools associate calls with modem resources based on the called-number (DNIS). In a mixed environment, where the server receives both modem and voice calls, you need to identify the service type of a call by using the **incoming called-number** command.

Without this, the server attempts to resolve whether an incoming call is a modem or voice call based on the interface over which the call comes. If the call comes in over an interface associated with a modem pool, the call is assumed to be a modem call; if a call comes in over a voice port associated with a dial peer, the call is assumed to be a voice call.

It helps to understand the logic behind the algorithm the system uses to distinguish voice and modem calls. The algorithm is as follows:

```
If the called-number matches a number from the modem pool,
 handle the call as a modem call
If the called-number matches a configured dial peer incoming called number,
 handle the call as a voice call
Else handle the call as a modem call by default modem pool
```

If there is no called-number information configured, call classification is handled as follows:

```
If the interface matches the interface configured for the modem pool,
 handle the call as a modem call.
If the voice port matches the one configured as the dial peer port,
 handle the call as a voice call
Else handle the call as a modem call by default modem pool
```

To identify the service type of a call to be voice, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	`Router(config)#` **`dial-peer voice`** *`number`* **`pots`**	Enter the dial peer configuration mode to configure a POTS peer.
2	`Router(config-dial-peer)#` **`incoming called-number`** *`number`*	Specify direct inward dial for this POTS peer.

Configuring VoIP Peers

VoIP peers enable outgoing calls to be made from a particular telephony device. To configure a VoIP peer, you need to uniquely identify the peer (by assigning it a unique tag number), define its destination telephone number and destination IP address. As with POTS peers, under most circumstances, the default values for the remaining dial peer configuration commands will be adequate to establish connections.

To configure a VoIP peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	`Router(config)#` **`dial-peer voice`** *`number`* **`voip`**	Enters the dial peer configuration mode to configure a VoIP peer. The number value of the dial-peer voice voip command is a tag that uniquely identifies the dial peer.
2	`Router(config-dial-peer)#` **`destination-pattern`** *`string`*	Defines the destination telephone number associated with this VoIP dial peer.
3	`Router(config-dial-peer)#` **`tech-prefix`** *`number`*	Specifies that a particular technology prefix be prepended to the destination patter of this dial peer.
4	`Router(config-dial-peer)#` **`session-target`** `{`**`ipv4:`***`destination-address`* `|` **`dns:`**`[`**`$s$.`**`|`**`$d$.`**`|`**`$e$.`**`|`**`$u$.`**`]` *`host-name`*`|`**`loopback:rtp`**`|`**`loopback:compressed`**`|` **`loopback:unompressed`**`|`**`ras`**`}`	Specifies a destination IP address for this dial peer.

For additional VoIP dial peer configuration options, refer to the "Voice-Related Commands" section of the Cisco IOS Release 12.0 *Voice, Video, and Home Applications Command Reference*, the Cisco IOS Release 12.0(3)T *Voice over IP for the Cisco AS5300* feature module, and the Cisco IOS Release 12.0(3)T *Service Provider Features for Voice over IP* feature module.

Verifying Dial Peer Configuration

- If you have relatively few dial peers configured, you can use the **show dial-peer voice** command to verify that the data configured is correct. Use this command to display a specific dial peer or to display all configured dial peers.
- Use the **show dialplan number** command to show the dial peer to which a particular number (destination pattern) resolves.

Troubleshooting Tips

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the chapter, "Configuring IP," in the Cisco IOS 11.3 *Network Protocols Configuration Guide, Part 1*.
- Use the **show dial-peer voice** command to verify that the operational status of the dial peer is up.

- Use the **show dialplan number** command on the local and remote routers to verify that the data is configured correctly on both.
- If you have configured number expansion, use the **show num-exp** command to check that the partial number on the local router maps to the correct full E.164 telephone number on the remote router.
- If you have configured a CODEC value, there can be a problem if both VoIP dial peers on either side of the connection have incompatible CODEC values. Make sure that both VoIP peers have been configured with the same CODEC value.
- Use the **debug voip ccani inout** command to verify the output string the router dials is correct.
- Use the **debug cch323 rtp** command to check RTP packet transport.
- Use the **debug cch323 h245** command to check logical channel negotiation.
- Use the **debug cch323 h225** command to check the call setup.

Configuring the Cisco AS5800 as an H.323 Gateway

The Service Provider implementation of Voice over IP uses both gatekeepers and gateways. Because of the extensive capabilities of the Cisco AS5800 universal access server, it is likely that it will function as a carrier class gateway in a Service Provider environment. The final step in configuring the Cisco AS5800 for Voice over IP functionality is to configure one of its interfaces as a gateway interface. You can use either an interface that is connected to the gatekeeper or a loopback interface for the gateway interface. The interface that is connected to the gatekeeper is usually a LAN interface—Fast Ethernet, Ethernet, FDDI, or Token Ring.

To configure a gateway interface, perform the following steps beginning in the global configuration mode:

Step	Command	Purpose
1	`Router(config)#` **`gateway`**	Enables the gateway.
2	`Router(config)#` **`ip cef`**	Enables Cisco Express Routing.
3		Configure the interface. This step will vary, depending on the interface you select as being the interface connected to the gatekeeper. For the purposes of this procedure, a Fast Ethernet interface is used.
4	`Router(config)#` **`int fa0`**	Enters configuration mode for the configured Fast Ethernet interface connected to the gatekeeper.
5	`Router(config-if)#` **`h323-gateway voip interface`**	Identifies this interface as a VoIP gateway interface.
6	`Router(config-if)#` **`h323-gateway voip id`** *`gatekeeper-id`* `{`**`ipaddr`** *`ip-address`* `[`*`port-number`*`]`|**`multicast`**`}`	Defines the name and location of the gatekeeper for this gateway.
7	`Router(config-if)#` **`h323-gateway voip h323-id`** *`interface-id`*	Defines the H.323 name of the gateway, identifying this gateway to its associated gatekeeper.
8	`Router(config-if)` **`h323-gateway voip tech-prefix`** *`prefix`*	Defines the technology prefix that the gateway will register with the gatekeeper.

For more information about configuring gateways and gatekeepers, refer to the Cisco IOS Release 12.0(3)T *Service Provider Features for Voice over IP* feature module.

Verifying Gateway Interface Configuration

Use the **show gateway** command to find the current registration information and status of the gateway.

Configuring the Cisco AS5800 for Interactive Voice Response

The Interactive Voice Response (IVR) Service Provider application provides IVR capabilities using Tool Command Language (TCL) scripts. For example, an IVR script is played when a caller receives a voice-prompt instruction to enter a specific type of information, such as a PIN. After playing the voice prompt, the IVR application collects the predetermined number of touch tones (digit collection) and forwards the collected digits to a server for storage and retrieval. Call records can be kept, and a variety of accounting functions performed.

Available IVR Scripts

The following is a description of the available IVR scripts:

- **fax_hop_on_1**—Collects digits from the redialer, such as account number and destination number. When placing the call to the H.323 network, the set of fields configured in the call information structure are *entered*, *destination*, and *account*.
- **clid_authen**—Authenticates the call with Automatic Number Identification (ANI) and Dialed Number Identification Service (DNIS), collects the destination data, and makes the call.
- **clid_authen_npw**—Same as **clid_authen**, but uses a null password when authenticating, rather than DNIS.
- **clid_authen_collect—**Authenticates the call with ANI and DNIS and collects the destination data, but if authentication fails, it collects the account and password.
- **clid_authen_col_npw**—Same as **clid_authen_collect**, but uses a null password and does not use or collect DNIS.
- **clid_col_npw_3**—Same as **clid_authen_col_npw** except if authentication with the digits collected (account and PIN number) failed, the script **clid_authen_col_npw** just played a failure message (auth_failed.au) and then hung up. This script, **clid_col_npw_3** allows two failures, then plays the retry audio file (auth_retry.au) and collects the account and PIN numbers again

  The caller can interrupt the message by entering digits for the account number which will trigger the prompt to enter the PIN number. If authentication fails the third time, the script plays the audio file **auth_fail_final.au**, then hangs up.

Configuring IVR

To use IVR with scripts, you need to configure the inbound POTS dial peer to support IVR, as well as enable IVR functionality by using the call application global configuration. To configure IVR, use the following commands beginning in the global configuration mode:

Step	Command	Purpose
1	`Router (config)#` **`call application voice`** *`name`*	Creates and then calls the application that interacts with the IVR feature.
2	`Router(config)#` **`dial-peer voice`** *`number`* **`pots`**	Enters the dial peer configuration mode to configure a POTS peer.
3	`Router(config-dial-peer)#` **`application`** *`name`*	Selects an IVR session application for the dial peer to use.

Step	Command	Purpose
4	`Router(config-dial-peer)#` **`destination-pattern`** *`string`*	Defines the telephone number associated with this POTS dial peer.
5	`Router(config-dial-peer)#` **`port`** *`shelf/slot/port`***`:D`**	Associates this POTS dial peer with a specific logical dial interface.
6	`Router(config-dial-peer)#` **`prefix`** *`string`*	(Optional) Specifies the prefix for this POTS dial peer. The **prefix** *string* value is sent to the telephony interface first, before the telephone number (destination pattern) associated with this dial peer is sent.

For more information about configuring IVR, refer to the Cisco IOS Release 12.0(7)T *Configuring Interactive Voice Response for Cisco Access Platforms* feature module.

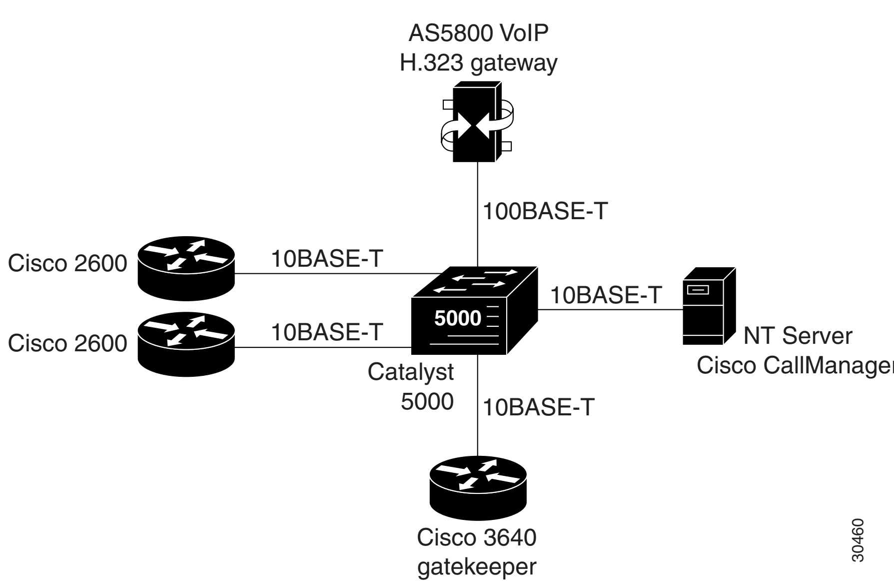
Verifying IVR Configuration

- If you have relatively few dial peers configured, you can use the **show dial-peer voice** command to verify that the data configured is correct. Use this command to display a specific dial peer or to display all configured dial peers.
- Use the **show running configuration** command to show all configured parameters relating to IVR.

Configuration Example

The following configuration example shows an abbreviated configuration using a Cisco 2600 router and a CiscoAS5800 universal access server as gateways and a Cisco 3600 router as a gatekeeper. Figure 6 shows the network diagram for this particular scenario.

Figure 6 AS5800 Universal Access Server Acting as a Gateway



Configuring the Cisco 3640 as a Gatekeeper

```
! Configure the Ethernet interface to be used at the gatekeeper interface.
interface Ethernet0/1
  ip address 172.30.00.00 255.255.255.0
  no ip directed-broadcast
  no logging event link-status
  no keepalive
!
! Configure the gatekeeper interface and enable the interface.
gatekeeper
  zone local gk3.gg-dn1 gg-dn1 173.50.00.00
  zone prefix gk3.gg-dn1 21*
  gw-type-prefix 9#* gw ipaddr 173.60.0.0 1720
  gw-type-prefix 6#* gw ipaddr 173.60.0.199 1720
  no use-proxy gk3.gg-dn1 default inbound-to terminal
  no shutdown
!
```

Configuring the Cisco 2600 as a Gateway

```
! Configure POTS and VoIP dial peers.
dial-peer voice 88 voip
  destination-pattern 11111
  tech-prefix 9#
  session ras
!
dial-peer voice 11 pots
  incoming called-number 11111
  destination-pattern 6#12345
port 1/1/1
prefix 12345
!
! Configure the gateway interface.
interface Ethernet0/0
ip address 173.60.0.199 255.255.255.0
no ip directed-broadcast
no ip mroute-cache
no logging event link-status
no keepalive
no cdp enabled
h323-gateway voip interface
h323-gateway voip id gk3.gg-dn1 ipaddr 173.30.0.0 1719
h323-gateway voip h323-id gw6@gg-dn1
h323-gateway voip tech-prefix 6#
!
```

Configuring the Cisco AS5800 as a Gateway

```
! Configure the T1 controller. (This configuration is for a T3 card.)
controller T1 1/0/0:1
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
! Configure POTS and VoIP dial peers.
dial-peer voice 11111 pots
 incoming called-number 12345
 destination-pattern 9#11111
 direct-inward-dial
 port 1/0/0:1:D
 prefix 11111
```

```
!
dial-peer voice 12345 voip
 destination-pattern 12345
 tech-prefix 6#
 session target ras
!
! Enable gateway functionality.
gateway
!
! Enable Cisco Express Forwarding.
ip cef
!
! Configure and enable the gateway interface.
interface FastEthernet0/3/0
 ip address 173.60.0.0.255.255.255.0
 no ip directed-broadcast
 no keepalive
 full-duplex
 no cdp enable
 h323-gateway voip interface
 h323-gateway voip id gk3.gg-dn1 ipaddr 173.30.0.0 1719
 h323-gateway voip h323-id gw3@gg-dn1
 h323-gateway voip tech-prefix 9#
!
! Configure the serial interface.(This configuration is for a T3 serial interface.)
interface Serial1/0/0:1:23
 no ip address
 no ip directed-broadcast
 ip mroute-cache
 isdn switch-type primary-5ess
 isdn incoming-voice modem
 no cdp enable
```

Command Reference

This section documents new or modified commands. All other commands used with this feature are documented in one of the following Cisco IOS documentation:

- Cisco IOS Release 12.0 *Voice, Video, and Home Applications Command Reference*
- Cisco IOS Release 12.0 *Dial Solutions Command Reference*
- Cisco IOS Release 12.0(3)T *Voice over IP for the AS5300* feature module
- Cisco IOS Release 12.0(3)T *Service Provider Features for Voice over IP* feature module
- Cisco IOS Release 12.0(7)T *Configuring Interactive Voice Response for Cisco Access Platforms* feature module

New Commands

- **dtmf-relay**
- **show vrm vdevice**
- **show vrm active_calls**
- **test vrm busyout**
- **test vrm reset**
- **test vrm unbusyout**

Modified Commands

- **codec**
- **port**
- **show csm**
- **show voice port**
- **voice-port**

In Cisco IOS Release 12.0(1)T or later, you can search and filter the output for **show** and **more** commands. This functionality is useful when you need to sort through large amounts of output, or if you want to exclude output that you do not need to see.

To use this functionality, enter a **show** or **more** command followed by the "pipe" character (**|**)**,** one of the keywords **begin**, **include**, or **exclude,** and an expression that you want to search or filter on:

command **|** {**begin** **|** **include** **|** **exclude**} *regular-expression*

Following is an example of the **show atm vc** command in which you want the command output to begin with the first line where the expression "PeakRate" appears:

show atm vc | begin PeakRate

For more information on the search and filter functionality, refer to the Cisco IOS Release 12.0(1)T feature module titled *CLI String Search*.

codec

To specify the voice coder rate of speech for a dial peer, use the **codec** dial-peer configuration command. To restore the default voice coder rate of speech value, use the **no** form of this command.

codec {**g711alaw** | **g711ulaw** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmfr**}

no codec

Syntax Description

g711alaw	G.711 A-Law 64000 bits per second (bps).
g711ulaw	G.711 u-Law 64000 bps.
g723r53	G.723.1 5300 bps.
g723r63	G.723.1 6300 bps.
g726r16	G.726 16000 bps.
g726r24	G.726 24000 bps.
g726r32	G.726 32000 bps.
g728	G.728 16000 bps.
g729abr8	G.729 ANNEX-A & B 8000 bps.
g729ar8	G.729 ANNEX-A 8000 bps.
g729br8	G.729 ANNEX-B 8000 bps.
g729r8	G.729 8000 bps.
gsmfr	GSMFR 13200 bps.

Defaults

g729r8.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(3)T	Support for Cisco 2600 series routers was added.
12.0(3)T	Support for the Cisco AS5300 access server was added.
12.0(7)T	Additional voice coder rates of speech were added.

Usage Guidelines

For toll quality, use the **g711alaw** or **g711ulaw** values. These values provide high-quality voice transmission but use a significant amount of bandwidth. For almost toll quality (and a significant savings in bandwidth), use the **g729r8** value.

If **codec** values for the VoIP peers of a connection do not match, the call will fail.

This command is only applicable to VoIP peers.

Examples

The following example configures a voice coder rate that provides toll quality but uses a relatively high amount of bandwidth:

```
dial-peer voice 10 voip
 codec g711alaw
```

Related Commands

Command	Description
dtmf-relay	Specifies how an H.323 gateway relays DTMF tones between telephony interfaces and an IP network.

dtmf-relay

To specify how an H.323 gateway relays dual tone multifrequency (DTMF) tones between telephony interfaces and an IP network, use the **dtmf-relay** dial-peer configuration command. To remove all signaling options and transmit the DTMF tones as part of the audio stream, use the **no** form of this command.

dtmf-relay [**cisco-rtp**] [**h245-alphanumeric**] [**h245-signal**]

no dtmf-relay

Syntax Description

cisco-rtp	(Optional) Forwards DTMF tones by using RTP protocol with a Cisco proprietary payload type.
h245-alphanumeric	(Optional) Forwards DTMF tones by using the H.245 "alphanumeric" User Input Indication method. Supports tones 0-9, *, #, and A-D.
h245-signal	(Optional) Forwards DTMF tones by using the H.245 "signal" User Input Indication method. Supports tones 0-9, *, #, and A-D.

Defaults

No default behavior or values.

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

DTMF is the tone generated when you press a digit on a touch-tone phone. This tone is compressed at one end of a call; when the tone is decompressed at the other end, it can become distorted, depending on the codec used. The DTMF relay feature transports DTMF tones generated after call establishment out of band using a standard H.323 out-of-band method and a proprietary RTP-based mechanism.

The gateway sends DTMF tones in the format you specify only if the remote device supports it. If the remote device supports multiple formats, the gateway chooses the format based on the following priority:

- cisco-rtp (highest priority)
- none, meaning that the DTMF is sent in-band

The principal advantage of the **dtmf-relay** command is that it transmits DTMF tones with greater fidelity than is possible in-band for most low-bandwidth CODECs, such as G.729 and G.723. Without the use of DTMF relay, calls established with low-bandwidth CODECs may have trouble accessing automated DTMF-based systems, such as voice-mail, menu-based ACD systems, and automated banking systems.

Note The **cisco-rtp** option of the **dtmf-relay** command is a proprietary Cisco implementation and only operates between two Cisco AS5800 universal access servers running Cisco IOS Release 12.0(2)XH, or between Cisco AS5800 universal access servers or Cisco 2600 or 3600 modular access routers running Cisco IOS Release 12.0(2)XH or later releases. Otherwise, the DTMF relay feature does not function, and the gateway sends DTMF tones in-band.

Examples

The following example configures DTMF relay with the **cisco-rtp** option when sending DTMF tones to dial-peer 103:

```
5800# configure terminal
5800(config)# dial-peer voice 103 voip
5800(config-dial-peer)# dtmf-relay cisco-rtp
5800(config-dial-peer)# end
5800#
```

The next example configures the gateway to send DTMF in-band (the default) when sending DTMF tones to dial-peer 103:

```
5800# configure terminal
5800(config)# dial-peer voice 103 voip
5800(config-dial-peer)# no dtmf-relay
5800(config-dial-peer)# end
```

Related Commands

Command	Description
codec	Specifies the voice coder rate of speech for a dial peer.

port

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

Cisco *2600/3600 Series Router*

port *slot/subunit/port*

no port

Cisco *MC3810*

port *slot/port*

no port

Cisco *AS5300 Access Server*

port *controller number***:D**

no port

Cisco *AS5800 Access Server*

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

no port

Syntax Description

*controller number***:D**	Specifies the T1 or E1 controller; **:D** indicates the D channel associated with ISDN PRI. Valid entries for the controller number variable is 0 to 3.
*shelf/slot/port***:D**	Specifies the T1 or E1 controller on the T1 card; **:D** indicates the D-channel associated with ISDN PRI. Valid entries for the *shelf* variable is 0 to 9999. Valid entries for the *slot* variable is 0 to 11. Valid entries for the *port* variable is 0 to 11.
*shelf/slot/parent***:***port***:D**	Specifies the T1 controller on the T3 card; **:D** indicates the D-channel associated with ISDN PRI. Valid entries for the *shelf* variable is 0 to 9999. Valid entries for the *slot* variable is 0 to 11. Valid entries for the *port* variable is 1 to 28. The value for the *parent* variable is always 0.
port	Specifies the voice port number. Valid entries are 0 or 1.
slot	Specifies the slot number where the voice interface card is installed. Valid entries are 0 or 1.
subunit	Specifies the subunit on the voice interface card in the router where the voice port is located. Valid entries are 0 or 1.

Default

No port is configured.

Command Mode

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced (Cisco 3600 series router).
11.3(3)T	Port-specific values for the Cisco 2600 were added.
11.3 MA	Port-specific values for the Cisco MC3810 were added.
12.0(3)T	Port-specific values for the Cisco AS5300 were added.
12.0(7)T	Port-specific values for the Cisco AS5800 were added.

Usage Guidelines

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

This command applies only to POTS peers.

Example

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

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

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

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

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

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

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

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

The following example associates a Cisco AS5800 POTS dial peer 10 with voice port 1/0/0:1:D (T3 card):

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

show csm

To display the call switching module (CSM) statistics for a particular or all DSP channels or for a specific modem or DSP channel, use the **show csm** privileged EXEC command.

Cisco *AS5300 Access Server*

show csm {**modem** [*slot***/***port* **|** *modem-group-number*] **|** **voice [***slot***/***dspm***/***dsp***/***dsp-channel*]}

Cisco *AS5800 Universal Access Server*

show csm voice [*shelf/slot/port*]

Syntax Description

modem	Specifies CSM call statistics for modems.
voice	Specifies CSM call statistics for DSP channels.
*slot***/***port*	(Optional) Specifies the location (and thereby the identity) of a specific modem.
modem-group-number	(Optional) Displays configuration for the dial peer identified by the argument *number*. Valid entries are any integers that identify a specific dial peer, from 1 to 32767.
slot/dspm/dsp/dsp-channel	(Optional) Identifies the location of a particular DSP channel.
shelf/slot/port	(Optional) Identifies the location of the voice interface card.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.
12.0(3)T	Port-specific values for the Cisco AS5300 were added.
12.0(7)T	Port-specific values for the Cisco AS5800 were added.

Usage Guidelines

This command shows the information related to CSM, which includes the DSP channel, the start time of the call, the end time of the call, and the channel on the controller used by the call.

Use the **show csm modem** command to display the CSM call statistic information for a specific modem, for a group of modems, or for all modems. If a *slot/port* argument is specified, then CSM call statistics are displayed for the specified modem. If the *modem-group-number* argument is specified, the CSM call statistics for all of the modems associated with that modem group are displayed. If no keyword is specified, CSM call statistics for all modems on the AS5300 are displayed.

Use the **show csm voice** command to display CSM statistics for a particular DSP channel. If the *slot/dspm/dsp/dsp-channel* or *shelf/slot/port* argument is specified, the CSM call statistics for calls using the identified DSP channel will be displayed. If no argument is specified, all CSM call statistics for all DSP channels will be displayed.

Examples

The following is sample output from the Cisco AS5300 for the **show csm voice** command:

```
Router# show csm voice 2/4/4/0
 slot 2, dspm 4, dsp 4, dsp channel 0,
 slot 2, port 56, tone, device_status(0x0002): VDEV_STATUS_ACTIVE_CALL.

csm_state(0x0406)=CSM_OC6_CONNECTED, csm_event_proc=0x600E2678, current call thru PRI
line
invalid_event_count=0, wdt_timeout_count=0
wdt_timestamp_started is not activated
wait_for_dialing:False, wait_for_bchan:False
pri_chnl=TDM_PRI_STREAM(s0, u0, c22), tdm_chnl=TDM_DSP_STREAM(s2, c27)
dchan_idb_start_index=0, dchan_idb_index=0, call_id=0xA003, bchan_num=22
csm_event=CSM_EVENT_ISDN_CONNECTED, cause=0x0000
ring_no_answer=0, ic_failure=0, ic_complete=0
dial_failure=0, oc_failure=0, oc_complete=3
oc_busy=0, oc_no_dial_tone=0, oc_dial_timeout=0
remote_link_disc=0, stat_busyout=0
oobp_failure=0
call_duration_started=00:06:53, call_duration_ended=00:00:00,
total_call_duration=00:00:44
The calling party phone number = 408
The called party phone number  = 5271086
total_free_rbs_timeslot = 0, total_busy_rbs_timeslot = 0,
total_dynamic_busy_rbs_timeslot = 0, total_static_busy_rbs_timeslot = 0,
total_sw56_rbs_timeslot = 0, total_sw56_rbs_static_bo_ts = 0,
total_free_isdn_channels = 21, total_busy_isdn_channels =
0,total_auto_busy_isdn_channels = 0,
min_free_device_threshold = 0
```

The following is sample output from the Cisco AS5800 for the **show csm voice** command:

```
5800# show csm voice 1/8/19
 shelf 1, slot 8, port 19
VDEV_INFO:slot 8, port 19
vdev_status(0x00000401):VDEV_STATUS_ACTIVE_CALL.VDEV_STATUS_HASLOCK.
csm_state(0x00000406)=CSM_OC6_CONNECTED, csm_event_proc=0x60868B8C, current
call thru PRI line
invalid_event_count=0, wdt_timeout_count=0
watchdog timer is not activated
wait_for_bchan:False
pri_chnl=(T1 1/0/0:22), vdev_chnl=(s8, c19)
start_chan_p=0, chan_p=62436D58, call_id=0x800D, bchan_num=22
The calling party phone number =
The called party phone number  = 7511
ring_no_answer=0, ic_failure=0, ic_complete=0
dial_failure=0, oc_failure=0, oc_complete=1
oc_busy=0, oc_no_dial_tone=0, oc_dial_timeout=0
remote_link_disc=0, busyout=0, modem_reset=0
call_duration_started=3d16h, call_duration_ended=00:00:00,
total_call_duration=00:00:00
```

Table 1 explains the fields contained in both of these examples.

Table 1 show csm voice Field Descriptions

Field	Description
slot	Indicates the slot where the VFC resides.
shelf/slot/port	Specifies the T1 or E1 controller.
dspm/dsp/dsp channel	Indicates which DSP channel is engaged in this call.
dsp	Indicates the DSP through which this call is established.
slot/port	This is the logical port number for the device. This is equivalent to the DSP channel number. The port number is derived from: (max_number_of_dsp_channels per dspm=12) * the dspm # (0-based) + (max_number_of_dsp_channels per dsp=2) * the dsp # (0-based) + the dsp channel number (0-based).
tone	Indicates which signalling tone is being used (DTMF, MF, R2). This only applies to CAS calls. Possible values are: — mf — dtmf — r2-compelled — r2-semi-compelled — r2-non-compelled

Table 1 show csm voice Field Descriptions (continued)

Field	Description
device_status	The status of the device. Possible values are: — VDEV_STATUS_UNLOCKED—Device is unlocked (meaning that it is available for new calls). — VDEV_STATUS_ACTIVE_WDT—Device is allocated for a call and the watchdog timer is set to time the connection response from the central office. — VDEV_STATUS_ACTIVE_CALL—Device is engaged in an active, connected call. — VDEV_STATUS_BUSYOUT_REQ—Device is requested to busyout; does not apply to voice devices. — VDEV_STATUS_BAD—Device is marked as bad and not usable for processing calls. — VDEV_STATUS_BACK2BACK_TEST—Modem is performing back-to-back testing (for modem calls only). — VDEV_STATUS_RESET—Modem needs to be reset (for modem only). — VDEV_STATUS_DOWNLOAD_FILE—Modem is downloading a file (for modem only). — VDEV_STATUS_DOWNLOAD_FAIL—Modem has failed during downloading a file (for modem only). — VDEV_STATUS_SHUTDOWN—Modem is not powered up (for modem only). — VDEV_STATUS_BUSY—Modem is busy (for modem only). — VDEV_STATUS_DOWNLOAD_REQ—Modem is requesting connection (for modem only).

Table 1 show csm voice Field Descriptions (continued)

Field	Description
csm_state	CSM call state of the current call (PRI line) associated with this device. Possible values are: — CSM_IDLE_STATE—Device is idle. — CSM_IC_STATE—A device has been assigned to an incoming call. — CSM_IC1_COLLECT_ADDR_INFO—A device has been selected to perform ANI/DNIS address collection for this call. ANI/DNIS address information collection is in progress. The ANI/DNIS is used to decide whether the call should be processed by a modem or a voice DSP. — CSM_IC2_RINGING—The device assigned to this incoming call has been told to get ready for the call. — CSM_IC3_WAIT_FOR_SWITCH_OVER—A new device is selected to take over this incoming call from the device collecting the ANI/DNIS address information. — CSM_IC4_WAIT_FOR_CARRIER—This call is waiting for the CONNECT message from the carrier. — CSM_IC5_CONNECTED—This incoming call is connected to the central office. — CSM_IC6_DISCONNECTING—This incoming call is waiting for a DISCONNECT message from the VTSP module to complete the disconnect process. — CSM_OC_STATE —An outgoing call is initiated. — CSM_OC1_REQUEST_DIGIT—The device is requesting the first digit for the dial-out number. — CSM_OC2_COLLECT_1ST_DIGIT—The first digit for the dial-out number has been collected. — CSM_OC3_COLLECT_ALL_DIGIT—All the digits for the dial-out number have been collected. — CSM_OC4_DIALING—This call is waiting for a dsx0 (B channel) to be available for dialing out. — CSM_OC5_WAIT_FOR_CARRIER—This (outgoing) call is waiting for the central office to connect. — CSM_OC6_CONNECTED—This (outgoing) call is connected. — CSM_OC7_BUSY_ERROR—A busy tone has been sent to the device (for VoIP call, no busy tone is sent; just a DISCONNECT INDICATION message is sent to the VTSP module) and this call is waiting for a DISCONNECT message from the VTSP module (or ONHOOK message from the modem) to complete the disconnect process. — CSM_OC8_DISCONNECTING—The central office has disconnected this (outgoing) call and the call is waiting for a DISCONNECT message from the VTSP module to complete the disconnect process.
csm_state: invalid_event_count=	Number of invalid events received by the CSM state machine.
wdt_timeout_count=	Number of times the watchdog timer is activated for this call.
wdt_timestamp_started	Indicates whether the watchdog timer is activated for this call.
wait_for_dialing:	Indicates whether this (outgoing) call is waiting for a free digit collector to become available to dial out the outgoing digits.

Table 1 show csm voice Field Descriptions (continued)

Field	Description
wait_for_bchan:	Indicates whether this (outgoing) call is waiting for a B channel to send the call out on.
pri_chnl=	Indicates which type of TDM stream is used for the PRI connection. For PRI and CAS calls, it will always be TDM_PRI_STREAM.
tdm_chnl=	Indicates which type of TDM stream is used for the connection to the device used to process this call. In the case of a VoIP call, this will always be set to TDM_DSP_STREAM.
dchan_idb_start_index=	First index to use when searching for the next IDB of a free D channel.
dchan_idb_index=	Index of the currently available IDB of a free D channel.
csm_event=	Event just passed to the CSM state machine.
cause	Event cause.
ring_no_answer=	Number of times call failed because there was no response.
ic_failure=	Number of failed incoming calls.
ic_complete=	Number of successful incoming calls.
dial_failure=	Number of times the connection failed because there was no dial tone.
oc_failure=	Number of failed outgoing calls.
oc_complete=	Number of successful outgoing calls.
oc_busy=	Number of outgoing calls where the connection failed because there was a busy signal.
oc_no_dial_tone=	Number of outgoing calls where the connection failed because there was no dial tone.
oc_dial_timeout=	Number of outgoing calls where the connection failed because the timeout value was exceeded.
call_duration_started=	Indicates the start of this call.
call_duration_ended=	Indicates the end of this call.
total_call_duration=	Indicates the duration of this call.
The calling party phone number =	Calling party number as given to CSM by ISDN.
The called party phone number =	Called party number as given to CSM by ISDN.
total_free_rbs_timeslot =	Total number of free RBS (CAS) timeslots available for the whole system.
total_busy_rbs_timeslot =	Total number of RBS (CAS) timeslots that have been busied out. This includes both dynamically and statically busied out RBS timeslots.
total_dynamic_busy_rbs_ timeslot =	Total number of RBS (CAS) timeslots that have been dynamically busied out.
total_static_busy_rbs_timeslot =	Total number of RBS (CAS) timeslots that have been statically busied out (that is, they are busied out using the CLI command)
total_free_isdn_channels =	Total number of free ISDN channels.
total_busy_isdn_channels =	Total number of busy ISDN channels.
total_auto_busy_isdn_channels =	Total number of ISDN channels that are automatically busied out.

Related Commands

Command	Description
show call active voice	Displays the Voice over IP active call table.
show call history voice	Displays the Voice over IP call history table.
show num-exp	Displays how the number expansions are configured in Voice over IP.
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** privileged EXEC command.

Cisco *2600/3600 Series Router*

show voice port *slot-number***/***subunit-number***/***port*

Cisco *MC3810*

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

Cisco *AS5300 Access Router*

show voice port *controller number***:D**

Cisco *AS5800 Universal Access Router*

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

Syntax Description

For the Cisco 2600/3600 series:

slot-number	Slot number in the Cisco router where the voice interface card is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed.
subunit-number	Subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1.
port	Voice port number. Valid entries are 0 or 1.

For the Cisco MC3810:

slot/port	(Optional) Displays information for only the voice port you specify with the *slot/port* designation. The *slot* variable specifies the slot number in the Cisco router where the voice interface card is installed. The only valid entry is 1. The *port* variable specifies the voice port number. Valid ranges are as follows: Analog voice ports: from 1 to 6. Digital voice port: Digital T1: from 1 to 24. Digital E1: from 1 to 15, and from 17 to 31.
summary	(Optional) Display a summary of all voice ports.

For the Cisco AS5300 Access Server:

controller number	Specifies the T1 or E1 controller.
:D	Indicates the D channel associated with ISDN PRI.

For the Cisco AS5800 Universal Access Server:

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

Command Mode

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3 MA	Port-specific values for the Cisco MC3810 were added.
12.0(3)T	Port-specific values for the Cisco AS5300 were added.
12.0(7)T	Port-specific values for the Cisco AS5800 were added.

Usage Guidelines

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

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

Examples

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

```
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 voice port on the Cisco 3600 series:

```
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 voice port on the 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 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 CODEC  VAD GAIN ATTN CANCEL
1/1  fxs-ls       up    up   on-hook    idle      729a    n    0    0    y
1/2  fxs-ls       up    up   on-hook    idle      729a    n    0    0    y
1/3  e&m-wnk      up    up   idle       idle      729a    n    0    0    y
1/4  e&m-wnk      up    up   idle       idle      729a    n    0    0    y
1/5  fxo-ls       up    up   idle       on-hook   729a    n    0    0    y
1/6  fxo-ls       up    up   idle       on-hook   729a    n    0    0    y
```

Table 2 explains the fields in the sample output.

Table 2 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.

Table 2 show voice port Field Descriptions (continued)

Field	Description
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.
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.

Table 2 show voice port Field Descriptions (continued)

Field	Description
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 that 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.

The following is sample output from the Cisco AS5800 for the **show voice port** command:

```
5800# show voice port 1/0/0:D
ISDN 1/0/0:D
 Type of VoicePort is ISDN
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is ""
 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 16 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 3 explains the fields in the sample output.

Table 3 show voice port Field Descriptions for the Cisco AS5800

Field	Description
Type of VoicePort	Indicates the voice port type.
Operational State	Operational state of the voice port.
Administrative State	Administrative state of the voice port.
Clear Wait Duration Timing	Time of inactive seizure signal to declare call cleared.
Currently Processing	Type of call currently being processed: none, voice, or fax.

Table 3 show voice port Field Descriptions for the Cisco AS5800 (continued)

Field	Description
Operations State	Operation state of the port.
Operation Type	Operation of the E&M signal: two-wire or four-wire.
Noise Regeneration	Whether or not background noise should be played to fill silent gaps if VAD is activated.
Non-Linear Processing	Whether or not non-linear processing is enabled for this port.
Music-On-Hold Threshold	Configured music-on-hold threshold value for this interface.
In Gain	Amount of gain inserted at the receiver side of the interface.
Out Attenuation	Amount of attenuation inserted at the transmit side of the interface.
Pulse Rate Timing	Pulse dialing rate in pulses per second (pps).
Echo Cancellation	Whether or not echo cancellation is enabled for this port.
Echo Cancel Coverage	Echo Cancel Coverage for this port.
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.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Regional Tone	Configured regional tone for this interface.

Related Commands

Command	Description
show call active voice	Displays the Voice over IP active call table.
show call history voice	Displays the Voice over IP call history table.
show dial-peer voice	Displays configuration information for dial peers.
show voice port	Displays configuration information about a specific voice port.

show vrm active_calls

To display active-only voice calls either for a specific VFC or all VFCs, use the **show vrm active_calls** privileged EXEC command.

show vrm active_calls {*dial-shelf-slot-number* | **all**}

Syntax Description

dial shelf slot number	Slot number of the dial shelf. Valid number is 0 to 13.
all	Lists all active calls for VFC slots.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use the **show vrm active_calls** to display active-only voice calls either for a specific VFC or all VFCs. Each active call occupies a block of information describing the call. This information provides basically the same information as the **show vrm vdevice** command.

Examples

The following is sample output from the **show vrm active_calls** command specifying dial shelf slot number:

```
5800# show vrm active_calls 6
slot =  6 virtual voice dev (tag) =  61 channel id = 2
capabilities list map = 9FFF
last/current codec loaded/used = None
TDM timeslot = 241
Resource (vdev_common) status = 401 means :active others
tot ingress data =  24
tot ingress control  = 1308
tot ingress data drops  = 0
tot ingress control drops  = 0
tot egress data  = 22051
tot egress control  = 1304
tot egress data drops  = 0
tot egress control drops  = 0
```

```
slot =  6 virtual voice dev (tag) =  40 channel id = 2
capabilities list map = 9FFF
last/current codec loaded/used = None
TDM timeslot = 157
Resource (vdev_common) status = 401 means :active others
```

Table 4 explains the fields in the sample output.

Table 4 show vrm vdevice Field Descriptions

Field	Description
slot	Slot where voice card is installed.
virtual voice dev (tag)	Identification number of the virtual voice device.
channel id	Identification number of the channel associated with this virtual voice device.
capability list map	Bitmaps for the codec supported on that DSP channel. Available values are: • CC_CAP_CODEC_G711U: 0x1 • CC_CAP_CODEC_G711A: 0x2 • CC_CAP_CODEC_G729IETF: 0x4 • CC_CAP_CODEC_G729a: 0x8 • CC_CAP_CODEC_G726r16: 0x10 • CC_CAP_CODEC_G726r24: 0x20 • CC_CAP_CODEC_G726r32: 0x40 • CC_CAP_CODEC_G728: 0x80 • CC_CAP_CODEC_G723r63: 0x100 • CC_CAP_CODEC_G723r53: 0x200 • CC_CAP_CODEC_GSM: 0x400 • CC_CAP_CODEC_G729b: 0x800 • CC_CAP_CODEC_G729ab: 0x1000 • CC_CAP_CODEC_G723ar63: 0x2000 • CC_CAP_CODEC_G723ar53: 0x4000 • CC_CAP_CODEC_G729: 0x8000
last/current codec loaded/used	Indicates the last codec loaded or used.
TDM timeslot	Time division multiplexing timeslot.
Resource (vdev_common) status	Current status of the VFC.
tot ingress data	Total amount of data (number of packets) sent from the PSTN side of the connection to the VoIP side of the connection.
tot ingress control	Total number of control packets sent from the PSTN side of the connection to the VoIP side of the connection.
tot ingress data drops	Total number of data packets dropped from the PSTN side of the connection to the VoIP side of the connection.
tot ingress control drops	Total number of control packets dropped from the PSTN side of the connection to the VoIP side of the connection.
tot egress data	Total amount of data (number of packets) sent from the VoIP side of the connection to the PSTN side of the connection.

Table 4 show vrm vdevice Field Descriptions (continued)

Field	Description
tot egress control	Total number of control packets sent from the VoIP side of the connection to the PSTN side of the connection.
tot egress data drops	Total number of data packets dropped from the VoIP side of the connection to the PSTN side of the connection.
tot egress control drops	Total number of control packets dropped from the VoIP side of the connection to the PSTN side of the connection.

Related Commands

Command	Description
show vrm vdevices	Displays detailed information for a specific DSP or a brief summary display for all VFCs.

show vrm vdevices

To display detailed information for a specific DSP or a brief summary display for all VFCs, use the **show vrm vdevices** privileged EXEC command.

show vrm vdevices {{*vfc-slot-number* | *voice-device-number*} | **summary**}

Syntax Description

vfc-slot-number	Slot number of the VFC. Valid number is 0 to 11.
voice-device-number	DSP number. Valid number is 1 to 96.
summary	List synopsis of voice feature card DSP mappings, capabilities, and resource states.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use the **show vrm vdevice** to display detailed information for a specific DSP or a brief summary display for all VFCs. The display provides information on the number of channels, channels per DSP, bitmap of DSPMs, version numbers, and so on. This information is useful in monitoring the current state of your VFCs.

The display for a specific DSP provides information on the codec that each channel is using, if active, or last used and if the channel is not currently transmitting cells. It also displays the state of the resource. In most cases, if there is an active call on that channel, the resource should be marked active. If the resource is marked as reset and/or bad, this may be an indication of a response loss for the VFC on a reset request. If this condition persists, you might experience a problem with the communication link between the router shelf and the VFC.

Examples

The following is sample output from the **show vrm vdevice** command specifying dial shelf slot number and DSP number. In this particular example, the call is active so the statistics displayed are for this active call. If no calls are currently active on the device, the statistics would be for the previous (or last active) call.

```
5800# show vrm vdevices 6 1
slot =  6 virtual voice dev (tag) =  1 channel id = 1
capabilities list map = 9FFF
last/current codec loaded/used = None
TDM timeslot = 0
Resource (vdev_common) status = 401 means :active others
tot ingress data =  101
tot ingress control  = 1194
tot ingress data drops  = 0
tot ingress control drops  = 0
tot egress data  = 39722
tot egress control  = 1209
tot egress data drops  = 0
tot egress control drops  = 0

slot =  6 virtual voice dev (tag) =  1 channel id = 2
capabilities list map = 9FFF
last/current codec loaded/used = None
TDM timeslot = 1
Resource (vdev_common) status = 401 means :active others
tot ingress data =  21
tot ingress control  = 1167
tot ingress data drops  = 0
tot ingress control drops  = 0
tot egress data  = 19476
tot egress control  = 1163
tot egress data drops  = 0
tot egress control drops  = 0
```

Table 5 explains the fields in the sample output.

Table 5 show vrm vdevice Field Descriptions

Field	Description
slot	Slot where voice card is installed.
virtual voice dev (tag)	Identification number of the virtual voice device.
channel id	Identification number of the channel associated with this virtual voice device.

Table 5 show vrm vdevice Field Descriptions (continued)

Field	Description
capability list map	Bitmaps for the codec supported on that DSP channel. Available values are: • CC_CAP_CODEC_G711U: 0x1 • CC_CAP_CODEC_G711A: 0x2 • CC_CAP_CODEC_G729IETF: 0x4 • CC_CAP_CODEC_G729a: 0x8 • CC_CAP_CODEC_G726r16: 0x10 • CC_CAP_CODEC_G726r24: 0x20 • CC_CAP_CODEC_G726r32: 0x40 • CC_CAP_CODEC_G728: 0x80 • CC_CAP_CODEC_G723r63: 0x100 • CC_CAP_CODEC_G723r53: 0x200 • CC_CAP_CODEC_GSM: 0x400 • CC_CAP_CODEC_G729b: 0x800 • CC_CAP_CODEC_G729ab: 0x1000 • CC_CAP_CODEC_G723ar63: 0x2000 • CC_CAP_CODEC_G723ar53: 0x4000 • CC_CAP_CODEC_G729: 0x8000
last/current codec loaded/used	Indicates the last codec loaded or used.
TDM timeslot	Time division multiplexing timeslot.
Resource (vdev_common) status	Current status of the VFC. Possible field values are: • FREE = 0x0000 • ACTIVE_CALL = 0x0001 • BUSYOUT_REQ = 0x0002 • BAD = 0x0004 • BACK2BACK_TEST = 0x0008 • RESET = 0x0010 • DOWNLOAD_FILE = 0x0020 • DOWNLOAD_FAIL = 0x0040 • SHUTDOWN = 0x0080 • BUSY = 0x0100 • OIR = 0x0200 • HASLOCK = 0x0400 /* vdev_pool has locked port */ • DOWNLOAD_REQ = 0x0800 • RECOVERY_REQ = 0x1000 • NEGOTIATED = 0x2000 • OOS = 0x4000
tot ingress data	Total amount of data (number of packets) sent from the PSTN side of the connection to the VoIP side of the connection.
tot ingress control	Total number of control packets sent from the PSTN side of the connection to the VoIP side of the connection.

Table 5 show vrm vdevice Field Descriptions (continued)

Field	Description
tot ingress data drops	Total number of data packets dropped from the PSTN side of the connection to the VoIP side of the connection.
tot ingress control drops	Total number of control packets dropped from the PSTN side of the connection to the VoIP side of the connection.
tot egress data	Total amount of data (number of packets) sent from the VoIP side of the connection to the PSTN side of the connection.
tot egress control	Total number of control packets sent from the VoIP side of the connection to the PSTN side of the connection.
tot egress data drops	Total number of data packets dropped from the VoIP side of the connection to the PSTN side of the connection.
tot egress control drops	Total number of control packets dropped from the VoIP side of the connection to the PSTN side of the connection.

The following is sample output from the **show vrm devices** command specifying a summary list. In the Voice Device Mapping area, the C_Ac column indicates number of active calls for a specific DSP. If there are any non zero numbers under the C_Rst and/or C_Bad column, this indicates a reset request was sent but it was lost; this could mean a faulty DSP.

```
5800# show vrm vdevices summary
***********************************************************
******summary of voice devices for all voice cards*********
***********************************************************

slot = 6 major ver = 0 minor ver = 1 core type used = 2
number of modules = 16 number of voice devices (DSPs) = 96
chans per vdevice = 2 tot chans = 192 tot active calls = 178
module presense bit map = FFFF tdm mode = 1 num_of_tdm_timeslots = 384
auto recovery is on

number of default voice file (core type images) = 2
file 0 maj ver = 0 min ver = 0 core_type = 1
trough size = 2880 slop value = 0 built-in codec bitmap = 0
loadable codec bitmap = 0 fax codec bitmap = 0

file 1 maj ver = 3 min ver = 1 core_type = 2
trough size = 2880 slop value = 1440 built-in codec bitmap = 40B
loadable codec bitmap = BFC fax codec bitmap = 7E


-------------------Voice Device Mapping------------------------
Logical Device (Tag)  Module#  DSP#  C_Ac  C_Busy  C_Rst  C_Bad
---------------------------------------------------------------
1                     1        1     2     0       0      0
2                     1        2     2     0       0      0
3                     1        3     2     0       0      0
4                     1        4     2     0       0      0
5                     1        5     2     0       0      0
6                     1        6     2     0       0      0
+++++++++++++++++++++++++++++++++++++++++++++++++++++++++++
7                     2        1     2     0       0      0
8                     2        2     2     0       0      0
9                     2        3     2     0       0      0
10                    2        4     1     0       0      0
11                    2        5     2     0       0      0
12                    2        6     1     0       0      0
+++++++++++++++++++++++++++++++++++++++++++++++++++++++++++
```

```
<information deleted>
++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++
91                    16       1     2     0       0       0
92                    16       2     2     0       0       0
93                    16       3     1     0       0       0
94                    16       4     2     0       0       0
95                    16       5     2     0       0       0
96                    16       6     2     0       0       0
++++++++++++++++++++++++++++++++++++++++++++++++++++++++++++

Total active call channels = 178
Total busied out channels = 0
Total channels in reset = 0
Total bad channels = 0
Note :Channels could be in multiple states
```

Table 6 explains the fields in the sample output.

Table 6 show vrm vdevice summary Field Descriptions

Field	Description
slot	Slot number where VFC is installed.
major ver	Major version of firmware running on VFC.
minor ver	Minor version of firmware running on VFC.
core type used	Type of DSPware in use. Possible field values are: • 1 = UBL (boot loader) • 2 = high complexity core • 3 = medium complexity core • 4 = low complexity core • 255 = invalid.
number of modules	Number of modules on the VFC. Maximum number possible is 16.
number of voice devices (DSP)s	Number of possible DSPs. Maximum number is 96.
chans per vdevice	Number of channels (meaning calls) each DSP can handle.
tot chans	Total number of channels.
tot active calls	Total number of active calls on this VFC.
module presense bit map	Indicates a 16-bit bitmap, each bit representing a module.
tdm mode	Time division multiplex bus mode. Possibe field values are: • 0 = VFC is in classic mode • 1 = VFC is in plus mode. This field should always be 1.
num_of_tdm_timeslots	Total number of calls that can be handled by the VFC.
auto recovery	Indicates whether auto recovery is enabled. When autorecovery is enabled, the VRM will try to recover a DSP by resetting it if, for some reason, the DSP stops responding.
number of default voice file (core type images)	Number of DSPware files in use.
maj ver	Major version of the DSPware in use.
min ver	Minor version of the DSPware in use.

Table 6 show vrm vdevice summary Field Descriptions (continued)

Field	Description
core type	Type of DSPware in use: Possible field values are: • 1 = boot loader • 2 = high complexity core • 3 = medium complexity core • 4 = low complexity core
trough size	This value indirectly represents the complexity of the DSPware in use.
slop value	This value indirectly represents the complexity of the DSPware in use.
built-in codec bitmap	Represents the bitmap of the codec built into the DSP firmware. Possible field values are: • CC_CAP_CODEC_G711U 0x0001 • CC_CAP_CODEC_G711A 0x0002 • CC_CAP_CODEC_G729IETF 0x0004 • CC_CAP_CODEC_G729a 0x0008 • CC_CAP_CODEC_G726r16 0x0010 • CC_CAP_CODEC_G726r24 0x0020 • CC_CAP_CODEC_G726r32 0x0040 • CC_CAP_CODEC_G728 0x0080 • CC_CAP_CODEC_G723r63 0x0100 • CC_CAP_CODEC_G723r53 0x0200 • CC_CAP_CODEC_GSM 0x0400 • CC_CAP_CODEC_G729b 0x0800 • CC_CAP_CODEC_G729ab 0x1000 • CC_CAP_CODEC_G723ar63 0x2000 • CC_CAP_CODEC_G723ar53 0x4000 • CC_CAP_CODEC_G729 0x8000

Table 6 show vrm vdevice summary Field Descriptions (continued)

Field	Description
loadable codec bitmap	Represents the loadable codec bitmap for the loadable CODECs. Possible field values are: • CC_CAP_CODEC_G711U = 0x0001 • CC_CAP_CODEC_G711A = 0x0002 • CC_CAP_CODEC_G729IETF = 0x0004 • CC_CAP_CODEC_G729a = 0x0008 • CC_CAP_CODEC_G726r16 = 0x0010 • CC_CAP_CODEC_G726r24 = 0x0020 • CC_CAP_CODEC_G726r32 = 0x0040 • CC_CAP_CODEC_G728 = 0x0080 • CC_CAP_CODEC_G723r63 = 0x0100 • CC_CAP_CODEC_G723r53 = 0x0200 • CC_CAP_CODEC_GSM = 0x0400 • CC_CAP_CODEC_G729b = 0x0800 • CC_CAP_CODEC_G729ab = 0x1000 • CC_CAP_CODEC_G723ar63 = 0x2000 • CC_CAP_CODEC_G723ar53 = 0x4000 • CC_CAP_CODEC_G729 = 0x8000
fax codec bitmap	Represents the fax codec bitmap. Possible field values are: • FAX_NONE = 0x1 • FAX_VOICE = 0x2 • FAX_144 = 0x4 • FAX_96 = 0x8 • FAX_72 = 0x10 • FAX_48 = 0x20 • FAX_24 = 0x40
Logical Device (Tag)	Tag number or the DSP number on that VFC.
Module #	Number identifying the module associated with a specific logical device.
DSP#	Number identifying the DSP on the VFC.
C_Ac	Number of active calls on identified DSP.
C_Busy	Number of busied-out channels associated with identified DSP.
C_Rst	Number of channels in the reset state associated with identified DSP.
C_Bad	Number of defective (“bad”) channels associated with identified DSP.
Total active call channels	Total number of active calls.
Total busied out channels	Total number of busied-out channels.
Total channels in reset	Total number of channels in reset state.
Total bad channels	Total number of defective channels.

Related Commands

Command	Description
show vrm active_calls	Displays active-only voice calls either for a specific VFC or all VFCs.

test vrm busyout

To busyout a specific DSP or channels on a specific DSP, use the **test vrm busyout** privileged EXEC command.

test vrm busyout *slot-number* {*first-dsp-number* **{***last-dsp-number* | **{channel** *number***}**} | **all**

Syntax Description

slot-number	Number identifing the slot where the VFC is installed. Values for this field are 0 to 11.
first-dsp-number	Specifies the first DSP in a range to be busied out. Each VFC holds 96 DSPs, so the value for this argument is 1 to 96.
last-dsp-number	Specifies the last DSP in a range to be busied out. Each VFC holds 96 DSPs, so the value for this argument is 1 to 96.
channel	(Optional) Specifies that a certain channel on the specified DSPs will be busied out.
number	Indicates the channel to be busied out. Values are 1 or 2.
all	Indicates that all 96 DSPs on the VFC installed in the defined slot will be busied out.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use the **test vrm busyout** command to busy out either one specific DSP or a range of DSPs on a specific VFC. In addition, you can use this comand to busyout a particular channel on a specified DSP or range of DSPs. To restore the activity of the busied-out DSP(s), use the **test vrm unbusyout** command.

Examples

The following example busies out all of the DSPs and associated channels for the VFC located in slot 4:

```
router# test vrm busyout 4 all
```

The following example busied out all of the channels from DSP1 to DSP3 for the VFC located in slot 4:

```
router# test vrm busyout 4 1 3
```

The following example busies out only channel 2 of DSP1 for the VFC located in slot 4:

```
router# test vrm busyout 4 1 channel 2
```

Related Commands

Command	Description
test vrm unbusyout	Restores activity to a busied-out DSP or busied-out channels on a DSP.

test vrm reset

To reset a particular DSP, use the **test vrm reset** privileged EXEC command.

test vrm reset {*slot-number dsp-number*}

Syntax Description

slot-number	Number identifing the slot where the VFC is installed.
dsp-number	Number identifying the DSP to be reset.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use the **test vrm reset** command to send a hard reset command to an identified DSP. When this command is used, any active calls on all channels associated with this DSP are dropped. Under most circumstances, you will never need to use this command.

Examples

The following example resets DSP 4 on the VFC installed in slot 2:

```
router# test vrm reset 4 2
 Resetting voice device may termiate active calls [confirm}
 Reset command sent to voice card 4 for voice device 2.
```

test vrm unbusyout

To restore activity to a busied-out DSP or busied-out channels on a DSP, use the **test vrm unbusyout** privileged EXEC command.

test vrm unbusyout *slot-number* {*first-dsp-number* **{***last-dsp-number* | **{channel** *number***}**} | **all**

Syntax Description

slot-number	Number identifing the slot where the VFC is installed. Values for this field are 0 to 11.
first-dsp-number	Specifies the first DSP in a range to be restored. Each VFC holds 96 DSPs, so the value for this argument is 1 to 96.
last-dsp-number	Specifies the last DSP in a range to be restored. Each VFC holds 96 DSPs, so the value for this argument is 1 to 96.
channel	(Optional) Specifies that a certain channel on the specified DSPs will be restored.
number	Indicates the channel to be restored. Values are 1 or 2.
all	Indicates that all 96 DSPs on the VFC installed in the defined slot will be restored.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use the **test vrm unbusyout** command to restore either one specific DSP or a range of DSPs on a specific VFC. In addition, you can use this comand to restore a particular channel on a specified DSP or range of DSPs. To busy out a DSP (or range of DSPs) or to busy out a particular channel, use the **test vrm busyout** command.

Examples

The following example restores the activity of all of the DSPs and associated channels for the VFC located in slot 4:

```
router# test vrm unbusyout 4 all
```

The following example restores the activity of all the channels on the DSP from DSP1 to DSP3 for the VFC located in slot 4:

```
router# test vrm unbusyout 4 1 3
```

The following example restores the activity of only channel 2 of DSP1 for the VFC located in slot 4:

```
router# test vrm unbusyout 4 1 channel 2
```

Related Commands

Command	Description
test vrm busyout	Busies out a DSP or busies out channels on a DSP.

voice-port

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

Cisco *2600/3600 Series Router*

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

Cisco *MC3810*

voice-port [*slot***/***port*] [**summary**]

Cisco *AS5300 Access Router*

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

Cisco *AS5800 Universal Access Router*

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

Syntax Description

For the Cisco 2600/3600 series:

slot-number	Slot number in the Cisco router where the voice interface card is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed.
subunit-number	Subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1.
port	Voice port number. Valid entries are 0 or 1.

For the Cisco MC3810:

*slot***/***port*	(Optional) Displays information for only the voice port you specify with the *slot/port* designation. The *slot* variable specifies the slot number in the Cisco router where the voice interface card is installed. The only valid entry is 1. The *port* variable specifies the voice port number. Valid ranges are as follows: Analog voice ports: from 1 to 6. Digital voice port: Digital T1: from 1 to 24. Digital E1: from 1 to 15, and from 17 to 31.
summary	(Optional) Display a summary of all voice ports.

For the Cisco AS5300 Access Server:

controller number	Specifies the T1 or E1 controller.
:D	Indicates the D channel associated with ISDN PRI.

For the Cisco AS5800 Universal Access Server:

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

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(3)T	Support for Cisco 2600 series routers was added.
12.0(3)T	Support for the Cisco AS5300 Access Server was added.
12.0(7)T	Support for the Cisco AS5800 Access Server was added.

Usage Guidelines

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

Examples

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

```
configure terminal
 voice-port 1/0/0
```

The following example accesses the voice-port configuration mode for digital voice port 24 on a Cisco MC3810 with a DVM installed:

```
configure terminal
 voice-port 1/24
```

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

```
configure terminal
 voice-port 1:D
```

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

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

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

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

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies a tag number for a dial peer.

Debug Commands

This section documents new or modified **debug** commands. All other commands used with this feature are documented in one of the follwing Cisco IOS documentation:

- Cisco IOS Release 12.0 *Voice, Video, and Home Applications Command Reference*
- Cisco IOS Release 12.0 *Dial Soluions Command Reference*
- Cisco IOS Release 12.0(3)T *Voice over IP for the AS5300* feature module
- Cisco IOS Release 12.0(3)T *Service Provider Features for Voice over IP* feature module

New Debug Commands

- **debug vrm control**
- **debug vrm error**
- **debug inout**

debug vrm control

To display all control messages sent to and received from the DSP, use the **debug vrm control** privileged EXEC command. To stop displaying DSP-specific control messages, use the **no** form of this command.

[**no**] **debug vrm control**

Syntax Description

There are no arguments or keywords used in this command.

Defaults

No default behavior or values.

Command History

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

Examples

The following example displays DSP-specific control messages going to the VRM:

```
*Nov 22 19:17:49.351: SEND CONTROL slot 4 tag 1 size C
*Nov 22 19:17:49.351: content : 0 0 0 1 0 8 0 1 0 4B 0 0 0 0 0

*Nov 22 19:17:49.351: SEND CONTROL slot 4 tag 1 size 14
*Nov 22 19:17:49.351: content : 0 0 0 1 0 10 0 1 0 4A 0 1 0 0 0

*Nov 22 19:17:49.351: SEND CONTROL slot 4 tag 1 size 1C
*Nov 22 19:17:49.351: content : 0 0 0 1 0 18 0 1 0 5C 0 2 0 2 0

*Nov 22 19:17:49.351: SEND CONTROL slot 4 tag 1 size 16
*Nov 22 19:17:49.351: content : 0 0 0 1 0 12 0 1 0 4C 0 3 0 1 0

*Nov 22 19:17:49.351: SEND CONTROL slot 4 tag 1 size E
*Nov 22 19:17:49.351: content : 0 0 0 1 0 A 0 1 0 42 0 4 0 0 0

*Nov 22 19:17:49.351: SEND CONTROL slot 4 tag 1 size 10
*Nov 22 19:17:49.351: content : 0 0 0 1 0 C 0 1 0 5B 0 5 0 0 0

*Nov 22 19:17:49.351: SEND CONTROL slot 4 tag 1 size E
*Nov 22 19:17:49.351: content : 0 0 0 1 0 A 0 1 0 4E 0 6 FF DA 0

*Nov 22 19:17:51.995: SEND CONTROL slot 4 tag 1 size C
*Nov 22 19:17:51.995: content : 0 0 0 1 0 8 0 1 0 44 0 7 FF DA 0

*Nov 22 19:17:51.995: SEND CONTROL slot 4 tag 1 size C
*Nov 22 19:17:51.995: content : 0 0 0 1 0 8 0 1 0 47 0 8 FF DA 0

*Nov 22 19:17:51.995: SEND CONTROL slot 4 tag 1 size C
*Nov 22 19:17:51.995: content : 0 0 0 1 0 8 0 1 0 44 0 9 FF DA 0

*Nov 22 19:17:51.995: SEND CONTROL slot 4 tag 1 size 1C
*Nov 22 19:17:51.995: content : 0 0 0 1 0 18 0 1 0 5C 0 A 0 2 0

*Nov 22 19:17:51.995: SEND CONTROL slot 4 tag 1 size 1C
```

```
*Nov 22 19:17:51.995: content : 0 0 0 1 0 18 0 1 0 49 0 B 0 1 0

*Nov 22 19:17:54.815: SEND CONTROL slot 4 tag 1 size E
*Nov 22 19:17:54.815: content : 0 0 0 1 0 A 0 1 0 53 0 1 0 0 0

*Nov 22 19:17:54.815: SEND CONTROL slot 4 tag 1 size E
*Nov 22 19:17:54.815: content : 0 0 0 1 0 A 0 1 0 54 0 1 0 0 0

*Nov 22 19:17:54.815: SEND CONTROL slot 4 tag 1 size E
*Nov 22 19:17:54.815: content : 0 0 0 1 0 A 0 1 0 57 0 1 0 0 0

*Nov 22 19:17:54.827: nip_voice_service_cb : Msg from DS slot 4 cmd = 196.
*Nov 22 19:17:54.827: RECEIVED CONTROL slot 4 tag 1 size 1C
*Nov 22 19:17:54.827: content : 0 0 0 1 0 18 0 1 0 C4 0 1 8F EA 9B
*Nov 22 19:17:54.827: DSP msg 196 received
*Nov 22 19:17:54.827: nip_voice_service_cb : Msg from DS slot 4 cmd = 197.
*Nov 22 19:17:54.827: RECEIVED CONTROL slot 4 tag 1 size 24
*Nov 22 19:17:54.827: content : 0 0 0 1 0 20 0 1 0 C5 0 1 0 0 0
*Nov 22 19:17:54.827: DSP msg 197 received
*Nov 22 19:17:54.827: nip_voice_service_cb : Msg from DS slot 4 cmd = 200.
*Nov 22 19:17:54.827: RECEIVED CONTROL slot 4 tag 1 size 34
*Nov 22 19:17:54.827: content : 0 0 0 1 0 30 0 1 0 C8 0 1 0 0 0
*Nov 22 19:17:54.827: DSP msg 200 received

*Nov 22 19:17:58.539: SEND CONTROL slot 4 tag 1 size E
*Nov 22 19:17:58.539: content : 0 0 0 1 0 A 0 1 0 53 0 1 0 0 0

*Nov 22 19:17:58.539: SEND CONTROL slot 4 tag 1 size E
*Nov 22 19:17:58.539: content : 0 0 0 1 0 A 0 1 0 54 0 1 0 0 0

*Nov 22 19:17:58.539: SEND CONTROL slot 4 tag 1 size E
*Nov 22 19:17:58.539: content : 0 0 0 1 0 A 0 1 0 57 0 1 0 0 0

*Nov 22 19:17:58.551: nip_voice_service_cb : Msg from DS slot 4 cmd = 196.
*Nov 22 19:17:58.555: RECEIVED CONTROL slot 4 tag 1 size 1C
*Nov 22 19:17:58.555: content : 0 0 0 1 0 18 0 1 0 C4 0 1 8F EA 9B
*Nov 22 19:17:58.555: DSP msg 196 received
*Nov 22 19:17:58.555: nip_voice_service_cb : Msg from DS slot 4 cmd = 197.
*Nov 22 19:17:58.555: RECEIVED CONTROL slot 4 tag 1 size 24
*Nov 22 19:17:58.555: content : 0 0 0 1 0 20 0 1 0 C5 0 1 0 0 0
*Nov 22 19:17:58.555: DSP msg 197 received
*Nov 22 19:17:58.555: nip_voice_service_cb : Msg from DS slot 4 cmd = 200.
*Nov 22 19:17:58.555: RECEIVED CONTROL slot 4 tag 1 size 34
*Nov 22 19:17:58.555: content : 0 0 0 1 0 30 0 1 0 C8 0 1 0 0 0
*Nov 22 19:17:58.555: DSP msg 200 received
*Nov 22 19:18:02.127: SEND CONTROL slot 4 tag 1 size C
*Nov 22 19:18:02.127: content : 0 0 0 1 0 8 0 1 0 47 0 C 0 0 0
```

Format of the Send messages is as follows:

```
SEND CONTROL slot <slot#> tag <tag#> size <size>
content : <x x x x> <x x> <x x> <x x> <x x> <x x x>
          tag#      len   chan  msg  proc  rtp_header
```

Format for the Receive messages is as follows:

```
nip_voice_service_cb : Msg from DS slot <slot#> cmd = <msg>.
RECEIVED CONTROL slot <slot#> tag <tag#> size <size>
content : 0 0 0 1 0 18 0 1 0 C4 0 1 8F EA 9B
content : <x x x x> <x x> <x x> <x x> <x x> <x x x>
          tag#       len   chan  msg   proc  rtp_header
DSP msg <msg> received
```

Table 7 describes the fields in previous example.

Table 7 debug vrm control Field Descriptions

Field	Description
tag#	DSP number.
len	Length of the packet from the RTP header (the next two bytes).
chan	Channel number (the next two bytes).
msg	Message ID number (the next two bytes).
proc	Process ID (the next two bytes).
rtp_header	First three bytes of the RTP header.

Related Commands

Command	Description
debug vrm error	Displays debug messages for all DSP-specific error messages going to the voice resource manager (VRM).
debug vrm inout	Displays debug messages for all DSP-specific messages going to and coming from the voice resource manager (VRM).

debug vrm error

To display all DSP-specific error messages going to the voice resource manager (VRM), use the **debug vrm error** privileged EXEC command. To stop displaying DSP-specific error messages, use the **no** form of this command.

[**no**] **debug vrm error**

Syntax Description

There are no arguments or keywords used in this command.

Defaults

No default behavior or values.

Command History

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

Examples

The following examples show some possible outputs from the **debug vrm error** command, displaying DS_specific error messages.

This example shows that an error occurred when sending data from the DSP to IP network (ingress direction):

```
- vrm_vtsp_send_ingress_data : fs_input failed
```

This error message shows that an error occurred when sending control message from the DSP to VTSP:

```
- vrm_vtsp_send_ingress_control : failed
```

This error message shows that there is no voice card present and a voice call is attempted:

```
- vrm_vtsp: No Voice Card ready yet.
```

This error message shows that no free resource is available, and a voice call is attempted:

```
- vrm_vtsp_open : vdev_common not available
```

This error message shows that there is already an active call on this channel, so abort:

```
- vrm_vtsp_open : vchan_instance already in use ABORT OPEN
```

The following messages show that the VTSP did a “dirty close” on a particular channel. “Dirty close” means that the DSP did not respond to the VTSP's request for the final statistics of the call.

```
- vrm_vtsp_open : cdb->dsp_info not NULL Abort OPEN
- vrm_vtsp_close failure no vtsp_cdb_ptr
- vrm_vtsp_close: without a dsp_info!
- vrm_vtsp_close : dirty close on tag <tag#> channel <chan#>
```

The following error mesaage describes the status of the DSP (virtual device):

```
- vrm_vtsp_close : vdev freed not locked. Status <value>
```

Possibe status values are as follows:

- ACTIVE_CALL = 0x0001
- BUSYOUT_REQ = 0x0002
- BAD = 0x0004
- BACK2BACK_TEST = 0x0008
- RESET = 0x0010
- DOWNLOAD_FILE = 0x0020
- DOWNLOAD_FAIL = 0x0040
- SHUTDOWN = 0x0080
- BUSY = 0x0100
- OIR = 0x0200
- HASLOCK = 0x0400 /* vdev_pool has locked port */
- DOWNLOAD_REQ = 0x0800
- RECOVERY_REQ = 0x1000
- NEGOTIATED = 0x2000
- OOS = 0x4000

The following error message shows that a "set_codec" command was issued, but the codec was not supported by the DSP:

```
- VTSP_FAIL: codec <value> not supported
```

Possible codec values are as follows:

- 0 = voipCodecG729,
- 1 = voipCodecG729a,
- 2 = voipCodecG726r16,
- 3 = voipCodecG726r24,
- 4 = voipCodecG726r32,
- 5 = voipCodecG711ulaw,
- 6 = voipCodecG711Alaw,
- 7 = voipCodecG728,
- 8 = voipCodecG723r63,
- 9 = voipCodecG723r53,
- 10 = voipCodecGSM,
- 11 = voipCodecG729b,
- 12 = voipCodecG729ab,
- 13 = voipCodecG723ar63,
- 14 = voipCodecG723ar53,
- 15 = voipCodecG729IETF

This error message shows that there is no buffer left in the pool for the VTSP to send a message to the DSP. <Number> int his output referst o the number of times the VRM ran out of buffer space.

```
- vrm_vtsp_get_packet: no buffers <number>
```

This error message notifies the VRM of a DSP alarm:

```
- vrm_vtsp_indicate_alarm : alarm_type <value> slot <slot#> tag <tag#> chan <chan#>
```

Possible values for the alarm are as follows:

- FATAL_ERROR = 0x01
- MEMORY_ERROR = 0x02
- BUFFER_ERROR = 0x04
- DOWNLOAD_ERROR = 0x08
- CHECKSUM_ERROR = 0x10

This eror message shows that the DSP sent a defective message:

```
- vrm msg offset too big tag <tag#> vchan <chan#>
```

Table 8 expains the field contained in the previous example.

Table 8 debug vrm error Field Descriptions

Field	Description
slot#	Slot in the Cisco AS5800 where the VFC is installed.
tag#	DSP number. Possible values for this field are 1 to 96.
chan#	Channel number. Possible values for this field are 1 and 2.

This error message indicates that an alarm message was received from the VFC/DSP and was successfully sent to the VTSP:

```
- vrm_msg_process_alarm_msg for <slot#>.<tag#>.<chan#> , state=<value>
```

Possible state values are as follows:

- 0 = RESET
- 1 = ADMINDOWN
- 2 = CORE_READY
- 3 = CODEC_READY
- 4 = VOICE_IDLE
- 5 = FAX_IDLE
- 6 = VOICE_READY
- 7 = FAX_READY
- 8= DTMF_READY

Related Commands

Command	Description
debug vrm control	Displays debug messages for all DSP-specific control messages going to the voice resource manager (VRM).
debug vrm inout	Displays debug messages for all DSP-specific messages going to and coming from the voice resource manager (VRM).

debug vrm inout

To display debug messages for all DSP-specific messages going to and coming from the voice resource manager (VRM), use the **debug vrm inout** privileged EXEC command. To stop displaying DSP-specific messages, use the **no** form of this command.

[**no**] **debug vrm inout**

Syntax Description

There are no arguments or keywords used in this command.

Defaults

No default behavior or values.

Command History

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

Examples

The following example displays DSP-specific messages going to the VRM when a call is made:

```
*Jun 17 13:02:41.495:vrm_vtsp_open :vtsp_cdb_ptr 623D2170
*Jun 17 13:02:41.495:vrm_vtsp_open :VTSP_SUCCESS
*Jun 17 13:02:41.535:vrm_vtsp_get_capabilities :vtsp_cdb_ptr 623D2170
*Jun 17 13:02:41.535:vrm_vtsp_get_capabilities :vtsp_cdb_ptr 623D2170
*Jun 17 13:02:41.535:vrm_vtsp_set_codec :vtsp_cdb_ptr 623D2170 new_codec 5
*Jun 17 13:02:41.535:VTSP_SUCCESS:Codec 5 was loaded already.
*Jun 17 13:02:41.767:vrm_vtsp_set_codec :vtsp_cdb_ptr 623D2170 new_codec 5
*Jun 17 13:02:41.767:VTSP_SUCCESS:Codec 5 was loaded already.
```

The following example displays DSP-specific messages going to the VRM when a call is complete:

```
*Jun 17 13:02:49.119:vrm_vtsp_close :vtsp_cdb_ptr 623D2170
*Jun 17 13:02:49.119:vrm_vtsp_close :0x2 close OK
```

Related Commands

Command	Description
debug vrm control	Displays debug messages for all DSP-specific control messages going to the voice resource manager (VRM).
debug vrm error	Displays debug messages for all DSP-specific error messages going to the voice resource manager (VRM).

Glossary

AAA—Authentication, Authorization, and Accounting. AAA is a suite of network security services that provide the primary framework through which access control can be set up on your Cisco router or access server.

ACOM—Term used in G.165, “General Characteristics of International Telephone Connections and International Telephone Circuits: Echo Cancellers.” ACOM is the combined loss achieved by the echo canceller, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.

a-law—A voice compression technique commonly used in Europe.

ANI—Answer Number Indication. The calling number (number of calling party).

ARQ—Admission request.

Call leg—A logical connection between the router and either a telephony endpoint over a bearer channel, or another endpoint using a session protocol.

CAS—Channel Associated Signaling. In E1 applications, timeslot 16 is used to transmit CAS information. Each frame’s timeslot 16 carries signaling information (ABCD bits) for two of the B channel timeslots.

CIR—Committed Information Rate. The average rate of information transfer a subscriber (for example, the network administrator) has stipulated for a Frame Relay PVC.

codec—coder-decoder. Device that typically uses pulse code modulation to transform analog signals into a digital bit stream and digital signals back into analog signals. In Voice over IP, it specifies the voice coder rate of speech for a dial peer.

Data Link Connection Identifier (DLCI)—Frame Relay virtual circuit number corresponding to a particular destination. The DLCI is part of the Frame Relay header and is usually 10 bits long.

Dial peer—An addressable call endpoint. In Voice over IP, there are two kinds of dial peers: POTS and VoIP.

DNS—Domain Name System used to address translation to convert H.323 IDs, URLs, or e-mail IDs to IP addresses. DNS is also used to assist in the location of remote gatekeepers and to reverse-map raw IP addresses to host names of administrative domains.

DNIS—Dialed number identification service. The destination number.

DS0—A 64-Kbps channel on an E1 or T1 WAN interface.

DSP—Digital Signal Processor.

DTMF—Dual tone multifrequency. Use of two simultaneous voice-band tones for dial (such as touch tone).

E.164—The international public telecommunications numbering plan. A standard set by ITU-T which addresses telephone numbers.

E1—Wide-area digital transmission scheme. E1 is the European equivalent of a T1 line. The E1’s higher clock rate (2.048 MHz) allows for 32 64-Kbps channels, which include one channel for framing and one channel for D-channel information.

E&M—Ear and mouth RBS signaling.

Endpoint—An H.323 terminal or gateway. An endpoint can call and be called. It generates and/or terminates the information stream.

FIFO—First-in, first-out. In data communication, FIFO refers to a buffering scheme where the first byte of data entering the buffer is the first byte retrieved by the CPU. In telephony, FIFO refers to a queuing scheme where the first calls received are the first calls processed.

Gatekeeper—A gatekeeper maintains a registry of devices in the multimedia network. The devices register with the gatekeeper at startup, and request admission to a call from the gatekeeper.

The gatekeeper is an H.323 entity on the LAN that provides address translation and control access to the LAN for H.323 terminals and gateways. The gatekeeper may provide other services to the H.323 terminals and gateways, such as bandwidth management and locating gateways.

Gateway—A gateway allows H.323 terminals to communicate with non-H.323 terminals by converting protocols. A gateway is the point at which a circuit-switched call is encoded and repackaged into IP packets.

An H.323 gateway is an endpoint on the LAN that provides real-time two-way communications between H.323 terminals on the LAN and other ITU-T terminals in the WAN, or to another H.323 gateway.

H.323—An International Telecommunication Union (ITU-T) standard that describes packet-based video, audio, and data conferencing. H.323 is an umbrella standard that describes the architecture of the conferencing system, and refers to a set of other standards (H.245, H.225.0, and Q.931) to describe its actual protocol.

H.323 RAS—Registration, admission, and status. The RAS signaling function performs registration, admissions, bandwidth changes, status and disengage procedures between the VoIP gateway and the gatekeeper.

HSRP—Hot Standby Routing Protocol. HSRP is a Cisco proprietary protocol which provides a redundancy mechanism when more than one router is connected to the same segment/subnet of an Ethernet/FDDI/Token Ring network.

ISDN—Integrated Services Digital Network. ISDN is a communications protocol, offered by telephone companies, that permits telephone networks to carry data, voice, and other traffic.

ITU-T—Telecommunication standardization sector of ITU.

IVR—Integrated voice response. A software feature that allows the use of one of several interactive voice response scripts during the call processing functionality.

LEC—Local exchange carrier.

LRQ—Location request.

MCU—Multipoint control unit

mu-law—**a-law**—A voice compression technique commonly used in North America.

Multicast—A process of transmitting PDUs from one source to many destinations. The actual mechanism (that is, IP multicast, multi-unicast, etc.) for this process may be different for LAN technologies.

Multilink PPP—Multilink Point-to-Point Protocol. This protocol is a method of splitting, recombining, and sequencing datagrams across multiple logical data links.

Multipoint-unicast—A process of transferring Protocol Data Units (PDUs) where an endpoint sends more than one copy of a media stream to different endpoints. This may be necessary in networks which do not support multicast.

node—An H.323 entity that uses RAS to communicate with the gatekeeper. For example, an endpoint such as a terminal, proxy, or gateway.

PDU—Protocol Data Units. Used by bridges to transfer connectivity information.

PBX—Private Branch Exchange. Privately-owned central switching office.

PLAR—Private Line Auto Ringdown. This type of service results in a call attempt to some particular remote endpoint when the local extension is taken off-key.

POTS—Plain Old Telephone Service. Basic telephone service supplying standard single line telephones, telephone lines, and access to the Public Switched Telephone Network.

POTS dial peer—Dial peer connected via a traditional telephony network. POTS peers point to a particular voice-port on a voice network device.

PRI—Primary Rate Interface. PRI is an ISDN interface to primary rate access. Primary rate access consists of a single 64 Kbps D channel plus 23 T1 or 30 E1 B channels for voice or data.

PSTN—Public Switched Telephone Network. PSTN refers to the local telephone company.

PVC—Permanent Virtual Circuit.

QoS—Quality of Service, which refers to the measure of service quality provided to the user.

RAS—Registration, Admission, and Status Protocol. This is the protocol that is used between endpoints and the gatekeeper to perform management functions.

RBS—Robbed Bit Signaling

RRQ—Registration request.

RSVP—Resource Reservation Protocol. This protocol supports the reservation of resources across an IP network.

T1—Digital WAN carrier facility. T1 transmits DS-1 formatted data at 1.544 Mbps through the telephone-switching network, using AMI or B8ZS coding. T1 is the North American equivalent of an E1 line.

TCL—Tool Command Language. An interpreted script language developed by Dr. John Ousterhout of the University of California, Berkeley, and now developed and maintained by Sun Microsystems Laboratories.

U-law—A companding technique commonly used in North America. U-law is standardized as a 64-Kbps codec in ITU-T G.711.

SPI—Service provider interface.

TDM—Time division multiplexing. Technique in which information from multiple channels can be allocated bandwidth on a single wire based on preassigned time slots. Bandwidth is allocated to each channel regardless of whether the station has data to transmit.

VoIP—Voice over IP. The ability to carry normal telephone-style voice over an IP-based internet with POTS-like functionality, reliability, and voice quality. VoIP is a blanket term which generally refers to Cisco's standards based (H.323, etc.) approach to IP voice traffic.

VoIP dial peer—Dial peer connected via a packet network; in the case of Voice over IP, this is an IP network. VoIP peers point to specific VoIP devices.

VTSP—Voice telephony service provider.

Zone—A collection of all terminals (tx), gateways (GW), and Multipoint Control Units (MCU) managed by a single gatekeeper (GK). A Zone includes at least one terminal, and may or may not include gateways or MCUs. A Zone has only one gatekeeper. A Zone may be independent of LAN topology and may be comprised of multiple LAN segments which are connected using routes or other devices.