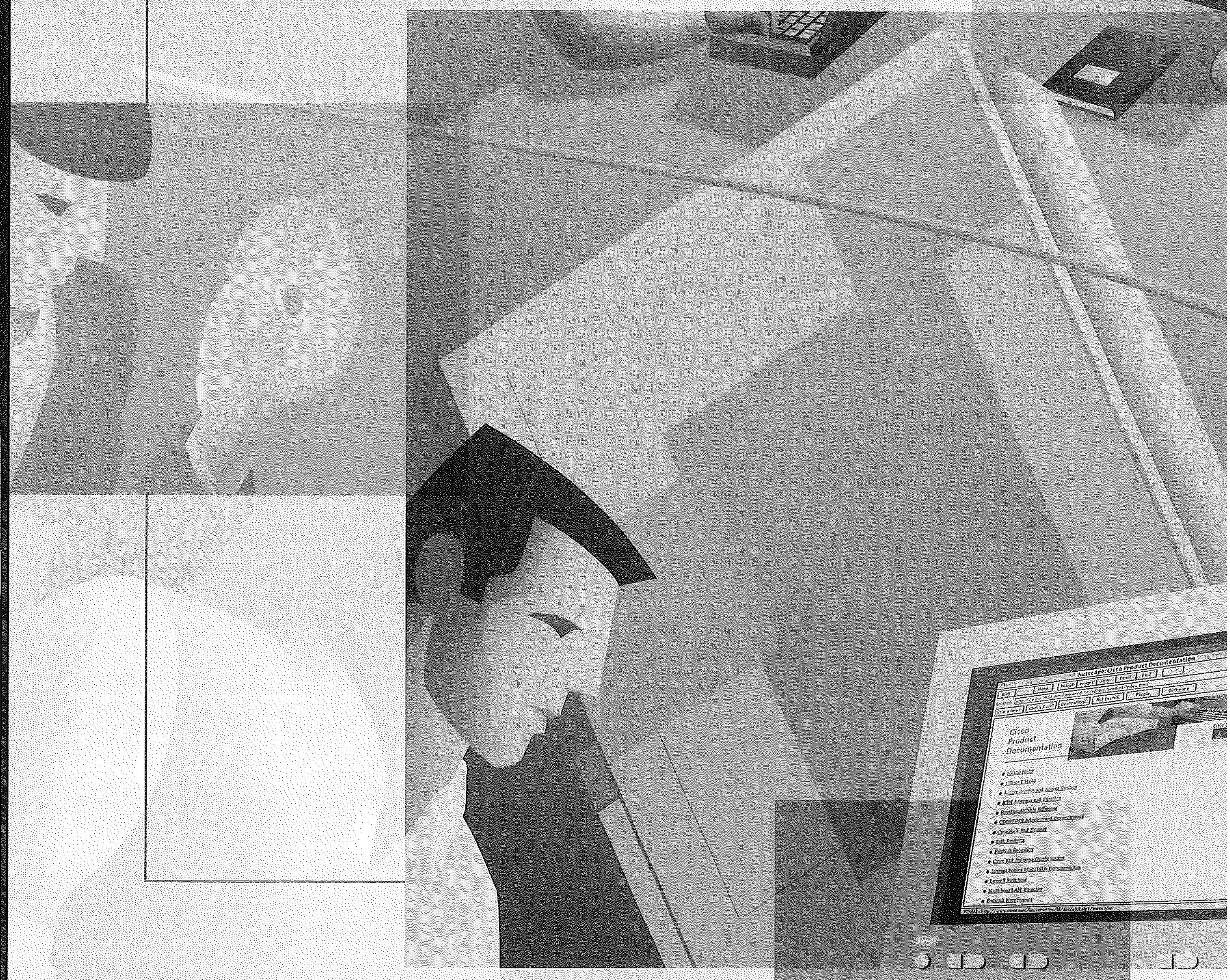
Cisco IOS Multiservice Applications Command Reference

Release 12.1



Documentation also available
on CD-ROM and the World Wide Web

Cisco Systems

Cisco Reader Comment Card

General Information

1 Years of networking experience ______ Years of experience with Cisco products ______

2 I have these network types: ☐ LAN ☐ Backbone ☐ WAN
☐ Other: ______

3 I have these Cisco products: ☐ Switches ☐ Routers
☐ Other: Specify model(s) ______

4 I perform these types of tasks: ☐ H/W Install and/or Maintenance ☐ S/W Config
☐ Network Management ☐ Other: ______

5 I use these types of documentation: ☐ H/W Install ☐ H/W Config ☐ S/W Config
☐ Command Reference ☐ Quick Reference ☐ Release Notes ☐ Online Help
☐ Other: ______

6 I access this information through: ____% Cisco Connection Online (CCO) ____% CD-ROM
____% Printed docs ____% Other: ______

7 Which method do you prefer? ______

8 I use the following three product features the most: ______

Document-specific Information

Document Title: Cisco IOS Multiservice Applications Command Reference

Part Number: 78-10259-01 S/W Release (if applicable): 12.1

On a scale of 1–5 (5 being the best) please let us know how we rate in the following areas:

______ The document was written at my technical level of understanding.

______ The information was accurate.

______ The document was complete.

______ The information I wanted was easy to find.

______ The information was well organized.

______ The information I found was useful to my job.

Please comment on our lowest score(s): ______

Mailing Information

Company Name ______ Date ______

Contact Name ______ Job Title ______

Mailing Address ______

City ______ State/Province ______ ZIP/Postal Code ______

Country ______ Phone ( ) ______ Extension ______

Fax ( ) ______ E-mail ______

Can we contact you further concerning our documentation? ☐ Yes ☐ No

You can also send us your comments by e-mail to **bug-doc@cisco.com**, or fax your comments to us at **(408) 527-8089**.

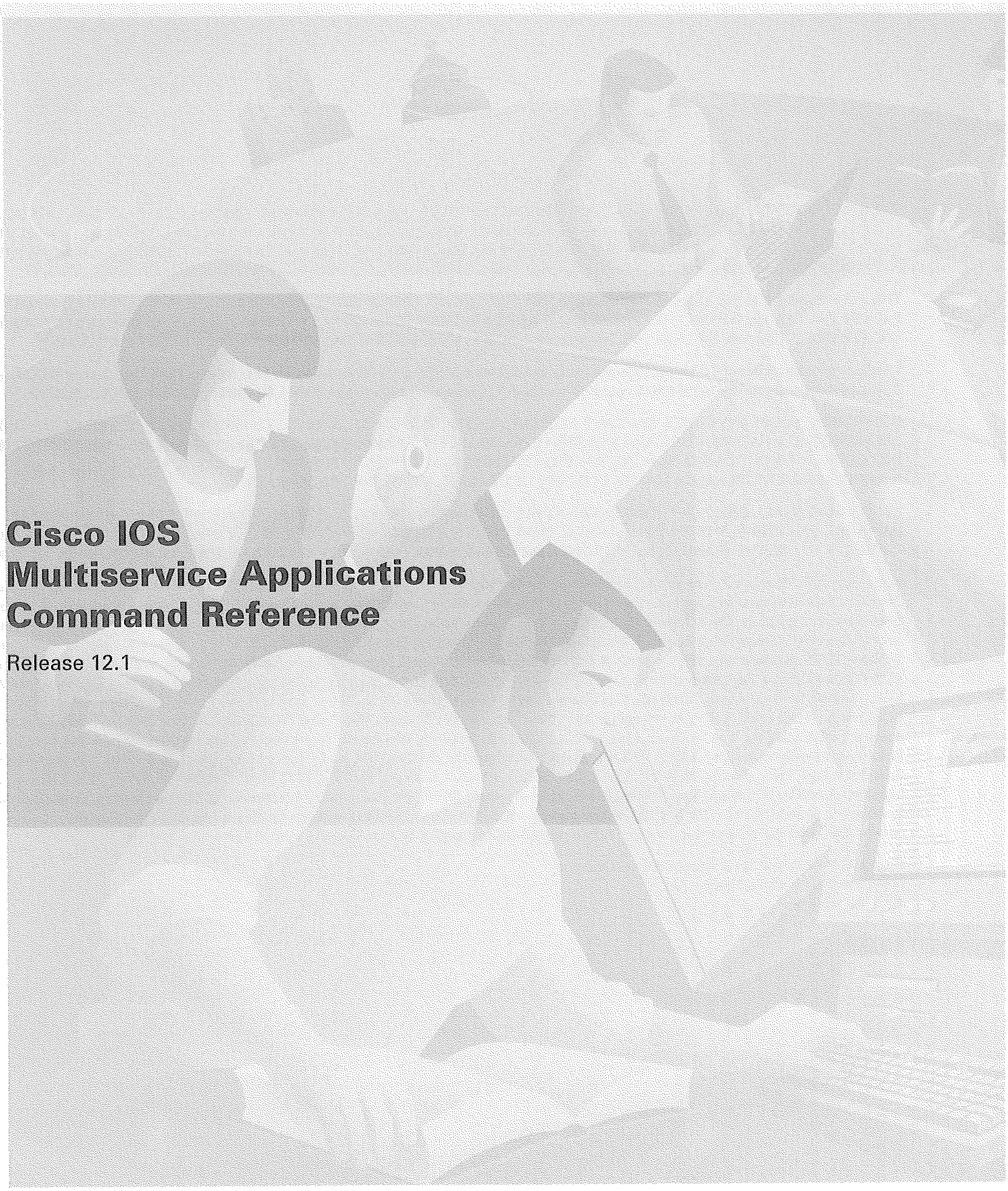
NO POSTAGE
NECESSARY
IF MAILED
IN THE
UNITED STATES

BUSINESS REPLY MAIL

FIRST-CLASS MAIL   PERMIT NO. 4631   SAN JOSE CA

POSTAGE WILL BE PAID BY ADDRESSEE

ATTN DOCUMENT RESOURCE CONNECTION
CISCO SYSTEMS INC
170 WEST TASMAN DRIVE
SAN JOSE CA 95134-9883

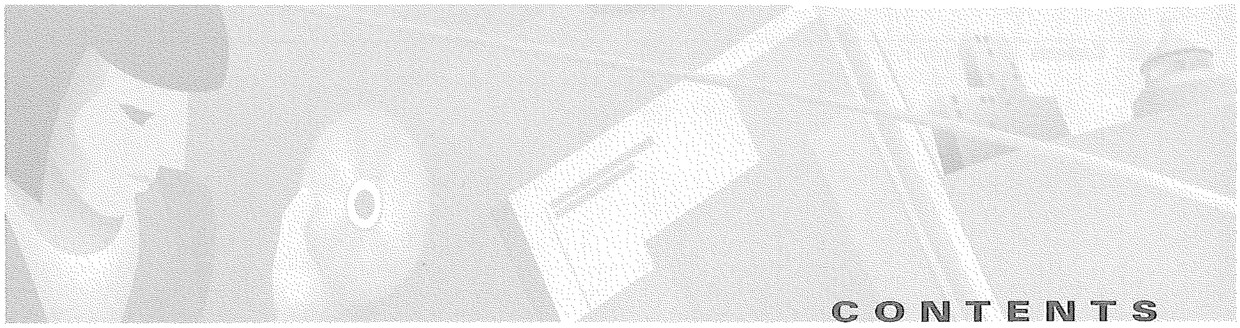


Cisco IOS Multiservice Applications Command Reference

Release 12.1

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

Customer Order Number: DOC-7810259=
Text Part Number: 78-10259-01



CONTENTS

About the Cisco IOS Software Documentation

This chapter discusses the objectives, audience, organization, and conventions of the Cisco IOS software documentation. It also discusses how to obtain documentation on Cisco Connection Online and the Documentation CD-ROM.

Documentation Objectives

This Cisco IOS software documentation describes the tasks and commands necessary to configure and maintain your networking device.

Audience

The Cisco IOS software documentation is intended primarily for users who configure and maintain networking devices, but are not necessarily familiar with tasks, the relationship between tasks, or the commands necessary to perform particular tasks.

Documentation Organization

The Cisco IOS software documentation is divided into 12 modules and 2 master indexes. In addition to the main documentation set, there are 4 supporting documents.

Documentation Modules

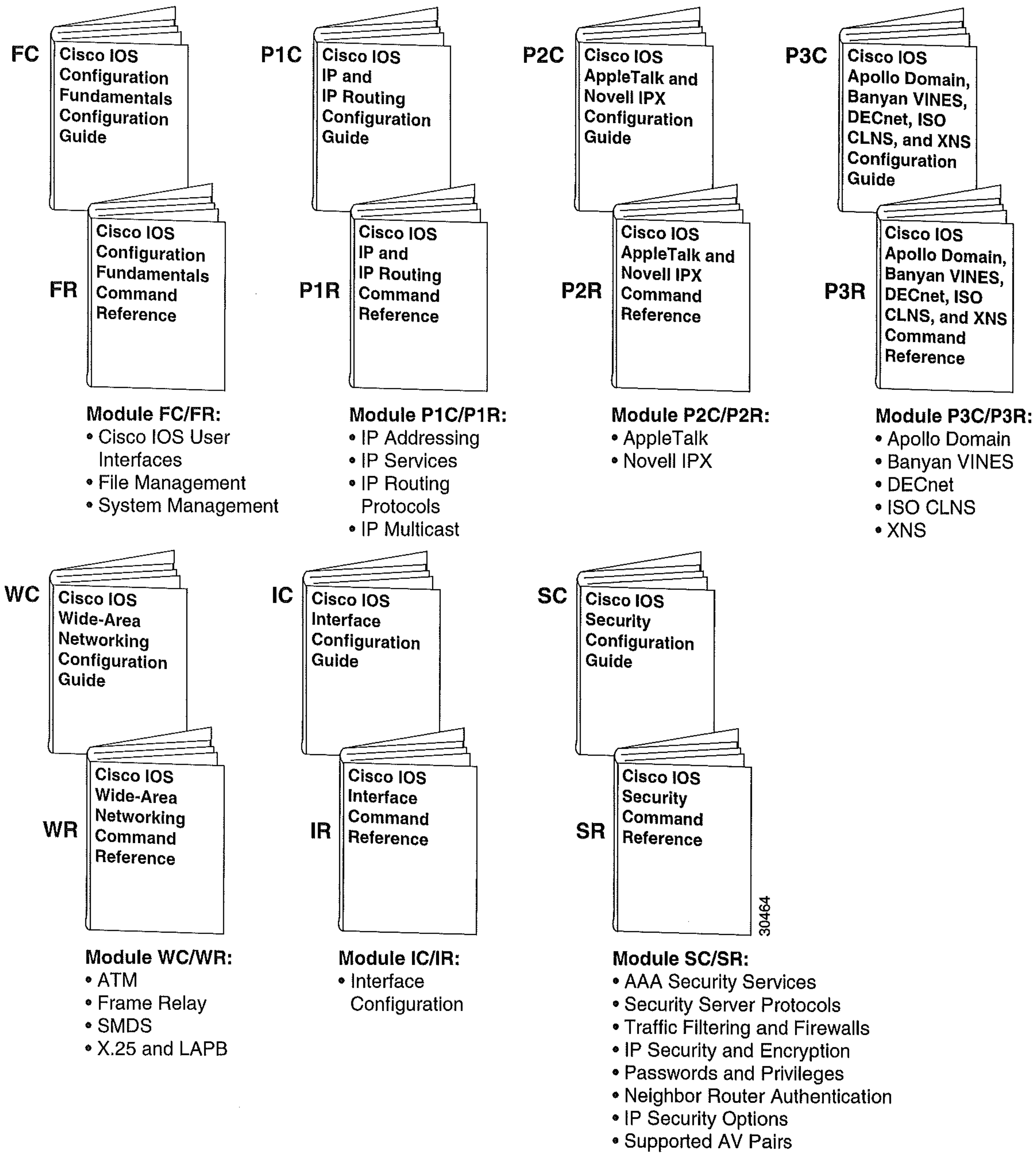
The Cisco IOS documentation modules consist of configuration guides and corresponding command reference publications. Chapters in a configuration guide describe protocols, configuration tasks, and Cisco IOS software functionality and contain comprehensive configuration examples. Chapters in a command reference publication provide complete command syntax information. Use each configuration guide in conjunction with its corresponding command reference publication.

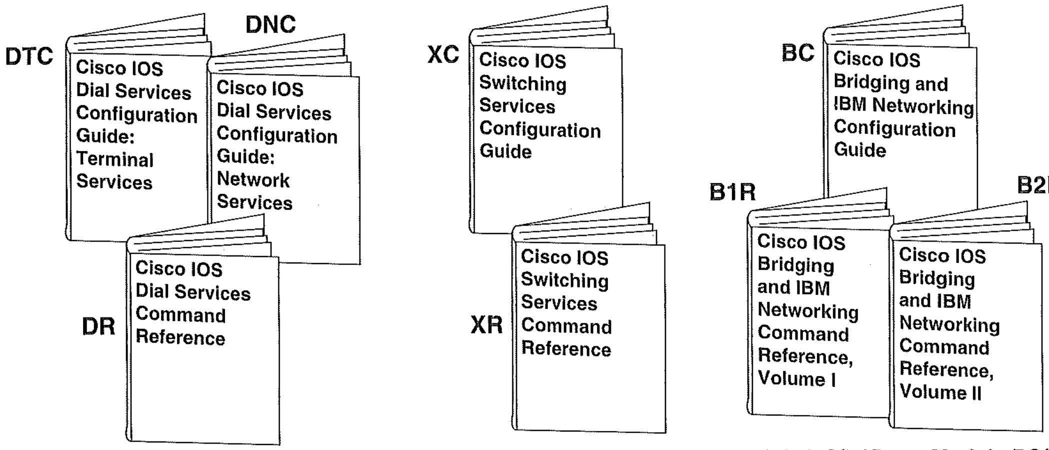
Documentation Set

The Cisco IOS software documentation set is shown in Figure 1.

Note The abbreviations next to the book icons are page designators (for example, FC, FR, and so on), which are defined in a key in the index of each document to help with navigation. The bulleted lists under each module describe the major technology areas discussed in their corresponding books.

Figure 1 Cisco IOS Software Documentation Modules





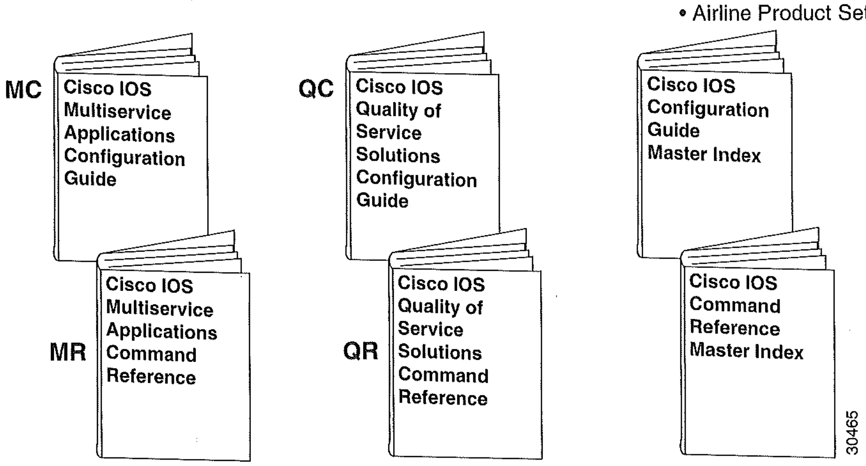
Module DTC/DR:
- Dial Access
- Modem Management
- ISDN BRI Services
- Point-to-Point Protocols
- Dial-on-Demand Routing
- Dial Backup
- Terminal Services

Module DNC/DR:
- Large-Scale Dial Solutions
- Cost-Control Solutions
- Virtual Private Networks
- X.25 on ISDN Solutions
- Telco Solutions
- Dial-Related Addressing Services
- Internetworking Dial Access Scenarios

Module XC/XR:
- Cisco IOS Switching Paths
- Cisco Express Forwarding
- NetFlow Switching
- Multiprotocol Label Switching
- Multilayer Switching
- Multicast Distributed Switching
- Virtual LANs
- LAN Emulation

Module BC/B1R:
- Transparent Bridging
- Source-Route Bridging
- Token Ring Inter-Switch Link
- Token Ring Route Switch Module
- Remote Source-Route Bridging
- Data-Link Switching Plus
- Serial Tunnel and Block Serial Tunnel
- LLC2 and SDLC
- IBM Network Media Translation
- SNA Frame Relay Access
- NCIA Client/Server
- Airline Product Set

Module BC/B2R:
- DSPU and SNA Service Point
- SNA Switching Services
- Cisco Transaction Connection
- Cisco Mainframe Channel Connection
- CLAW and TCP/IP Offload
- CSNA, CMPC, and CMPC+
- TN3270 Server



Module MC/MR:
- Voice over IP
- Voice over Frame Relay
- Voice over ATM
- Voice over HDLC
- Video Support
- Universal Broadband Features

Module QC/QR:
- Packet Classification
- Congestion Management
- Congestion Avoidance
- Policing and Shaping
- Signalling
- Link Efficiency Mechanisms

Master Indexes

Two master indexes provide indexing information for the Cisco IOS software documentation set: an index for the configuration guides, and an index for the command references. In addition, individual books contain a book-specific index.

Supporting Documents

The following documents support the Cisco IOS software documentation set:

- *Cisco IOS Command Summary*
- *Cisco IOS System Error Messages*
- *Cisco IOS Debug Command Reference*
- *Cisco IOS Dial Services Quick Configuration Guide*

Document Conventions

The Cisco IOS documentation set uses the following conventions:

Convention	Description
^ or Ctrl	^ or Ctrl represents the Control key. For example, the key combination ^D or Ctrl-D means hold down the Control key while you press the D key. Keys are indicated in capital letters but are not case sensitive.
string	A string is a nonquoted set of characters. For example, when setting an SNMP community string to public, do not use quotation marks around the string or the string will include the quotation marks.

Examples use the following conventions:

Convention	Description
`screen`	Courier plain shows an example of information displayed on the screen.
`boldface screen`	Courier bold shows an example of text that you must enter.
< >	Angle brackets show nonprinting characters, such as passwords.
!	An exclamation point at the beginning of a line indicates a comment line. (Exclamation points are also displayed by the Cisco IOS software for certain processes.)
[ ]	Square brackets show default responses to system prompts.

The following conventions are used to attract the attention of the reader:

Caution Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

Note Means *reader take note*. Notes contain helpful suggestions or references to materials not contained in this manual.

Timesaver Means the *described action saves time*. You can save time by performing the action described in the paragraph.

Within the Cisco IOS software documentation, the term *router* is generally used to refer to a variety of networking devices (for example, routers, access servers, and Route Switch Modules). Within examples, routers, access servers, and other networking devices that support Cisco IOS software are shown alternately. These products are used only for example purposes; that is, an example that shows one product does not indicate that other products are not supported.

Command Syntax Conventions

Command descriptions use the following conventions:

Convention	Description
boldface	Boldface text indicates commands and keywords that you enter literally as shown.
italics	Italic text indicates arguments for which you supply values.
[x]	Square brackets indicate an optional element (keyword or argument).
{x}	Braces indicate a required element (keyword or argument).
[x {y | z}]	Braces and vertical lines within square brackets indicate a required choice within an optional element.

Cisco Connection Online

Cisco Connection Online (CCO) is the primary, real-time support channel for Cisco Systems. Maintenance customers and partners can self-register on CCO to obtain additional information and services.

Available 24 hours a day, 7 days a week, CCO provides a wealth of standard and value-added services to customers and business partners of Cisco Systems. CCO services include product information, product documentation, software updates, release notes, technical tips, the Bug Navigator, configuration notes, brochures, descriptions of service offerings, and download access to public and authorized files.

CCO serves a wide variety of users through two interfaces that are updated and enhanced simultaneously: a character-based version and a multimedia version that resides on the World Wide Web (WWW). The character-based CCO supports Zmodem, Kermit, Xmodem, FTP, and Internet e-mail, and it is excellent for quick access to information over lower bandwidths. The WWW version of CCO provides richly formatted documents with photographs, figures, graphics, and video, as well as hyperlinks to related information.

You can access CCO in the following ways:

- http://www.cisco.com
- http://www-europe.cisco.com
- http://www-china.cisco.com
- Telnet: cco.cisco.com
- Modem: From North America, 408 526-8070; from Europe, 33 1 64 46 40 82. Use the following terminal settings: VT100 emulation; databits: 8; parity: none; stop bits: 1; and connection rates up to 28.8 kbps.

For a copy of the CCO Frequently Asked Questions (FAQ), contact cco-help@cisco.com. For additional information, contact cco-team@cisco.com.

Note If you are a network administrator and need personal technical assistance with a Cisco product that is under warranty or covered by a maintenance contract, contact the Cisco Technical Assistance Center (TAC) at 800 553-2447, 408 526-7209, or tac@cisco.com. To obtain general information about Cisco Systems, Cisco products, or upgrades, contact 800 553-6387, 408 526-7208, or cs-rep@cisco.com.

Documentation CD-ROM

Cisco documentation and additional literature are available in a CD-ROM package, which ships with your product. The Documentation CD-ROM, a member of the Cisco Connection Family, is updated monthly; therefore, it might be more current than printed documentation. To order additional copies of the Documentation CD-ROM, contact your local sales representative or call customer service. The CD-ROM package is available as a single package or as an annual subscription. You can also access Cisco documentation on the World Wide Web at http://www.cisco.com, http://www-china.cisco.com, or http://www-europe.cisco.com.

Providing Documentation Feedback

If you are reading Cisco product documentation on the World Wide Web, you can submit comments electronically. Click **Feedback** in the toolbar and select **Documentation**. After you complete the form, click **Submit** to send it to Cisco.

You can also submit feedback on Cisco documentation as follows:

- Mail in the Cisco Reader Comment Card located at the front of this book
- Send an e-mail to bug-doc@cisco.com
- Send a fax to 408 527-8089

We appreciate your comments.

Using Cisco IOS Software

This chapter provides helpful tips for understanding and configuring Cisco IOS software using the command-line interface (CLI). It contains the following sections:

- Understanding Command Modes
- Getting Help
- Using the No and Default Forms of Commands
- Saving Configuration Changes
- Searching and Filtering Output of show and more Commands

For an overview of Cisco IOS software configuration, refer to the *Cisco IOS Configuration Fundamentals Configuration Guide*.

For information on the conventions used in the Cisco IOS documentation set, see the "About the Cisco IOS Software Documentation" chapter located at the beginning of this book.

Understanding Command Modes

The Cisco IOS user interface is divided into many different modes. The commands available to you at any given time depend on which mode you are currently in. Entering a question mark (?) at the system prompt allows you to obtain a list of commands available for each command mode.

When you log in to the Cisco IOS software, you begin in user mode, often called EXEC mode. Only a limited subset of the commands are available in EXEC mode. To have access to all commands, you must enter privileged EXEC mode. Normally, you must enter a password to enter privileged EXEC mode. From privileged mode, you can enter any EXEC command or enter global configuration mode. Most of the EXEC commands are one-time commands, such as **show** commands, which show important status information, and **clear** commands, which clear counters or interfaces. The EXEC commands are not saved when the networking device reboots.

The configuration modes allow you to make changes to the running configuration. If you later save the configuration to the startup configuration, these commands are stored when the networking device reboots. To enter the various configuration modes, you must start at global configuration mode. From global configuration mode, you can enter interface configuration mode, subinterface configuration mode, and a variety of protocol-specific modes.

ROM monitor mode is a separate mode used when a networking device running Cisco IOS software cannot boot properly. If your networking device does not find a valid system image when it is booting, or if its configuration file is corrupted at startup, the system might enter ROM monitor mode.

Summary of Main Command Modes

Table 1 summarizes the main command modes of the Cisco IOS software.

Table 1 Summary of Main Command Modes

Command Mode	Access Method	Prompt	Exit Method
User EXEC	Log in.	`Router>`	Use the **logout** command.
Privileged EXEC	From user EXEC mode, use the **enable** EXEC command.	`Router#`	To exit back to user EXEC mode, use the **disable** command. To enter global configuration mode, use the **configure terminal** privileged EXEC command.
Global configuration	From privileged EXEC mode, use the **configure terminal** privileged EXEC command.	`Router(config)#`	To exit to privileged EXEC mode, use the **exit** or **end** command or press **Ctrl-Z**. To enter interface configuration mode, use an **interface** configuration command.
Interface configuration	From global configuration mode, enter by specifying an interface with an **interface** command.	`Router(config-if)#`	To exit to global configuration mode, use the **exit** command. To exit to privileged EXEC mode, use the **exit** command or press **Ctrl-Z**. To enter subinterface configuration mode, specify a subinterface with the **interface** command.
Subinterface configuration	From interface configuration mode, specify a subinterface with an **interface** command.	`Router(config-subif)#`	To exit to global configuration mode, use the **exit** command. To enter privileged EXEC mode, use the **end** command or press **Ctrl-Z**.
ROM monitor	From privileged EXEC mode, use the **reload** EXEC command. Press the Break key during the first 60 seconds while the system is booting.	`>`	To exit to user EXEC mode, use the **continue** command.

For more information regarding command modes, refer to the "Using the Command-Line Interface" chapter in the *Cisco IOS Configuration Fundamentals Configuration Guide*.

Getting Help

Entering a question mark (**?**) at the system prompt displays a list of commands available for each command mode. You can also get a list of keywords and arguments associated with any command by using the context-sensitive help feature.

To get help specific to a command mode, a command, a keyword, or an argument, use one of the following commands:

Command	Purpose
help	Obtains a brief description of the help system in any command mode.
abbreviated-command-entry?	Obtains a list of commands that begin with a particular character string. (No space between command and question mark.)
abbreviated-command-entry<**Tab**>	Completes a partial command name.
?	Lists all commands available for a particular command mode.
command ?	Lists the keywords or arguments that you must enter next on the command line. (Space between command and question mark.)

Example: How to Find Command Options

This section provides an example of how to display syntax for a command. The syntax can consist of optional or required keywords and arguments. To display keywords and arguments for a command, enter a question mark (?) at the configuration prompt, or after entering part of a command followed by a space. The Cisco IOS software displays a list of keywords and arguments available along with a brief description of them. For example, if you were in global configuration mode, typed the command **arap**, and wanted to see all the keywords or arguments that may be entered next on the command line, you would type **arap ?**.

Table 2 shows examples of how you can use the question mark (?) to assist you in entering commands. The table steps you through configuring a serial interface IP address on a Cisco 7206 router running Cisco IOS Release 12.0(3).

Table 2 How to Find Command Options

Command	Comment

`Router> enable`
`Password: <password>`
`Router#`
	Enter the **enable** command and password to access privileged EXEC commands.
	You are in privileged EXEC mode when the prompt changes to `Router#`.

`Router# configure terminal`
`Enter configuration commands, one per line. End with CNTL/Z.`
`Router(config)#`
	Enter the **configure terminal** privileged EXEC command to enter global configuration mode.
	You are in global configuration mode when the prompt changes to `Router(config)#`.

Table 2 How to Find Command Options (continued)

<table>
<tr><th>Command</th><th>Comment</th></tr>
<tr><td><pre>
Router(config)# interface serial ?
  <0-6>   Serial interface number
Router(config)# interface serial 4 ?
  /
Router(config)# interface serial 4/ ?
  <0-3>   Serial interface number
Router(config)# interface serial 4/0
Router(config-if)#
</pre></td><td>Enter interface configuration mode by specifying the serial interface that you want to configure using the interface serial global configuration command.

Enter a ? to display what you must enter next on the command line. In this example, you must enter the serial interface slot number and port number, separated by a back slash.

You are in interface configuration mode when the prompt changes to <code>Router(config-if)#</code>.</td></tr>
<tr><td><pre>
Router(config-if)# ?
Interface configuration commands:
  ...
  ip   Interface Internet Protocol config commands
  keepalive   Enable keepalive
  lan-name   LAN Name command
  llc2   LLC2 Interface Subcommands
  load-interval   Specify interval for load calculation for an
                       interface
  locaddr-priority   Assign a priority group
  logging   Configure logging for interface
  loopback   Configure internal loopback on an interface
  mac-address   Manually set interface MAC address
  mls   mls router sub/interface commands
  mpoa   MPOA interface configuration commands
  mtu   Set the interface Maximum Transmission Unit (MTU)
  netbios   Use a defined NETBIOS access list or enable
                       name-caching
  no   Negate a command or set its defaults
  nrzi-encoding   Enable use of NRZI encoding
  ntp   Configure NTP
  ...
Router(config-if)#
</pre></td><td>Enter a ? to display a list of all the interface configuration commands available for the serial interface. This example shows only some of the interface configuration commands that are available.</td></tr>
<tr><td><pre>
Router(config-if)# ip ?
Interface IP configuration subcommands:
  access-group   Specify access control for packets
  accounting   Enable IP accounting on this interface
  address   Set the IP address of an interface
  authentication   authentication subcommands
  bandwidth-percent   Set EIGRP bandwidth limit
  broadcast-address   Set the broadcast address of an interface
  cgmp   Enable/disable CGMP
  directed-broadcast   Enable forwarding of directed broadcasts
  dvmrp   DVMRP interface commands
  hello-interval   Configures IP-EIGRP hello interval
  helper-address   Specify a destination address for UDP broadcasts
  hold-time   Configures IP-EIGRP hold time
  ...
Router(config-if)# ip
</pre></td><td>Enter the command that you want to configure for the interface. In this example, the ip command is used.

Enter a ? to display what you must enter next on the command line. This example shows only some of the interface IP configuration subcommands that are available.</td></tr>
</table>

Table 2 How to Find Command Options (continued)

Command	Comment

Router(config-if)# **ip address ?**
  A.B.C.D   IP address
  negotiated   IP Address negotiated over PPP
Router(config-if)# **ip address**
	Enter the subcommand that you want to configure for the interface. In this example, the **address** subcommand is entered.
	
	Enter a **?** to display what you must enter next on the command line. In this example, you must enter an IP address or the **negotiated** keyword.
	
	Because a carriage return (<cr>) is not displayed, it indicates that you must enter more keywords or arguments to complete the command.

Router(config-if)# **ip address 172.16.0.1 ?** A.B.C.D IP subnet mask Router(config-if)# **ip address 172.16.0.1**	Enter the keyword or argument you want to use. In this example, the 172.16.0.1 IP address is entered. Enter a **?** to display what you must enter next on the command line. In this example, you must enter an IP subnet mask. Because a <cr> is not displayed, it indicates that you must enter more keywords or arguments to complete the command.
Router(config-if)# **ip address 172.16.0.1 255.255.255.0 ?** secondary Make this IP address a secondary address <cr> Router(config-if)# **ip address 172.16.0.1 255.255.255.0**	Enter the IP subnet mask. In this example, the 255.255.255.0 IP subnet mask is entered. Enter a **?** to display what you must enter next on the command line. In this example, you can enter the **secondary** keyword or press **Enter**. Because a <cr> is displayed, it indicates that you can press **Enter** to complete the command.
Router(config-if)# **ip address 172.16.0.1 255.255.255.0** Router(config-if)#	In this example, **Enter** is pressed to complete the command.

Using the No and Default Forms of Commands

Almost every configuration command also has a **no** form. In general, use the **no** form to disable a function. Use the command without the keyword **no** to reenable a disabled function or to enable a function that is disabled by default. For example, IP routing is enabled by default. To disable IP routing, use the **no ip routing** command and specify **ip routing** to reenable it. The Cisco IOS software command reference publications provide the complete syntax for the configuration commands and describe what the **no** form of a command does.

Configuration commands can also have a **default** form. The **default** form of a command returns the command setting to its default. Most commands are disabled by default, so the **default** form is the same as the **no** form. However, some commands are enabled by default and have variables set to certain default values. In these cases, the **default** command enables the command and sets variables to their default values. The Cisco IOS software command reference publications describe what the **default** form of a command does if the command is not the same as the **no** form.

Saving Configuration Changes

Enter the **copy system:running-config nvram:startup-config** command to save your configuration changes to your startup configuration so that they will not be lost if there is a system reload or power outage. For example:

```
Router# copy system:running-config nvram:startup-config
Building configuration...
```

It might take a minute or two to save the configuration. After the configuration has been saved, the following output appears:

```
[OK]
Router#
```

On most platforms, this task saves the configuration to NVRAM. On the Class A Flash file system platforms, this task saves the configuration to the location specified by the CONFIG_FILE environment variable. The CONFIG_FILE variable defaults to NVRAM.

Searching and Filtering Output of show and more Commands

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.

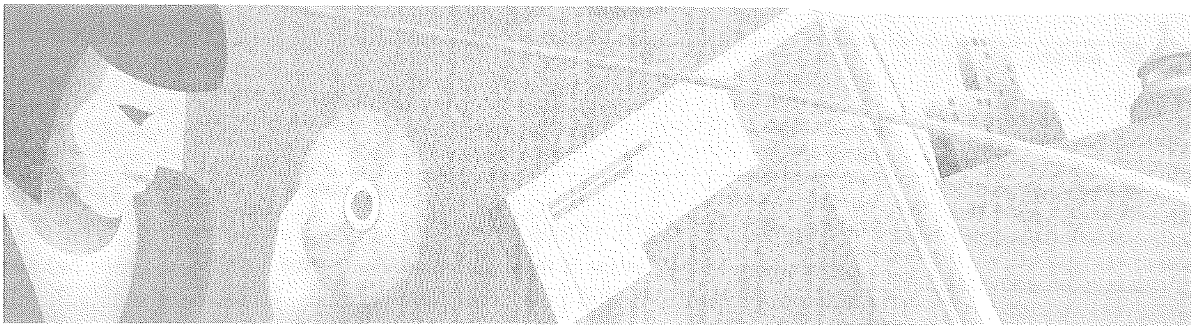
To use this functionality, enter a **show** or **more** command followed by the "pipe" character (|), one of the keywords **begin**, **include**, or **exclude**, and an expression that you want to search or filter on:

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

The following is an example of the **show interface** command in which you want the output to only include lines where the expression "protocol" appears:

```
Router# show interface | include protocol
FastEthernet0/0 is up, line protocol is up
Serial4/0 is up, line protocol is up
Serial4/1 is up, line protocol is up
Serial4/2 is administratively down, line protocol is down
Serial4/3 is administratively down, line protocol is down
```

For more information on the search and filter functionality, refer to the "Using the Command-Line Interface" chapter in the *Cisco IOS Configuration Fundamentals Configuration Guide*.



Multiservice Applications Commands: A through Ca

This book documents commands used to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features. Commands in this book are listed alphabetically. For information on how to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features, refer to the *Cisco IOS Multiservice Applications Configuration Guide*.

acc-qos

To generate an SNMP event if the quality of service for a dial peer drops below a specified level, use the **acc-qos** command in dial-peer configuration mode. To restore the default value, use the **no** form of this command.

acc-qos {best-effort | controlled-load | guaranteed-delay}

no acc-qos

Syntax Description

best-effort	Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation. This is the default.
controlled-load	Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to ensure that preferential service is received even when the bandwidth is overloaded.
guaranteed-delay	Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.

Defaults

best-effort

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command is only applicable to VoIP dial peers.

Use the **acc-qos** dial-peer configuration command to generate an SNMP event if the quality of service for specified dial peer drops below the specified level. When a dial peer is used, the Cisco IOS software reserves a certain amount of bandwidth so that the selected quality of service can be provided. Cisco IOS software uses RSVP to request quality of service guarantees from the network.

To select the most appropriate value for this command, you need to be familiar with the amount of traffic this connection supports and what kind of impact you are willing to have on it. The Cisco IOS software generates a trap message when the bandwidth required to provide the selected quality of service is not available.

Examples

The following example selects guaranteed-delay as the specified level below which an SNMP trap message will be generated:

```
dial-peer voice 10 voip
 acc-qos guaranteed-delay
```

Related Commands

Command	Description
req-qos	Specifies the desired quality of service to be used in reaching a specified dial peer in Voice over IP.

alias static

To create a static entry in the local alias table, use the **alias static** command in gatekeeper configuration mode. To remove a static entry, use the **no** form of this command.

alias static *ip-signalling-addr* [*port*] **gkid** *gatekeeper-name* [**ras** *ip-ras-addr port*] [**terminal** | **mcu** | **gateway** {**h320** | **h323-proxy** | **voip**}] [**e164** *e164-address*] [**h323id** *h323-id*]

no alias static *ip-signalling-addr* [*port*] **gkid** *gatekeeper-name* [**ras** *ip-ras-addr port*] [**terminal** | **mcu** | **gateway** {**h320** | **h323-proxy** | **voip**}] [**e164** *e164-address*] [**h323id** *h323-id*]

Syntax Description

ip-signalling-addr	IP address of the H.323 node, used as the address to signal when establishing a call.
[*port*]	(Optional) Port number other than the endpoint Call Signalling well-known port number (1720).
gkid *gatekeeper-name*	Name of the local gatekeeper whose zone this node is a member of.
ras *ip-ras-addr*	(Optional) Node RAS signalling address. If omitted, the *ip-signalling-addr* parameter is used in conjunction with the RAS well-known port.
port	(Optional) Port number other than the RAS well-known port number (1719).
terminal	(Optional) Indicates that the alias refers to a terminal.
mcu	(Optional) Indicates that the alias refers to an MCU.
gateway	(Optional) Indicates that the alias refers to a gateway.
h320	(Optional) Indicates that the alias refers to an H.320 node.
h323-proxy	(Optional) Indicates that the alias refers to an H.323 proxy.
voip	(Optional) Indicates that the alias refers to VoIP.
e164 *e164-address*	(Optional) Specifies the node E.164 address. This keyword and argument can be used more than once to specify as many E.164 addresses as needed. Note that there is a maximum number of 128 characters that can be entered for this address. To avoid exceeding this limit, you can enter multiple **alias static** commands with the same call signalling address and different aliases.
h323id *h323-id*	(Optional) Specifies the node H.323 alias. This keyword and argument can be used more than once to specify as many H.323 ID aliases as needed. Note that there is a maximum number of 256 characters that can be entered for this address. To avoid exceeding this limit, you can enter multiple **alias static** commands with the same call signalling address and different aliases.

Defaults

No static aliases exist.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Usage Guidelines

The local alias table can be used to load static entries by performing as many of the commands as necessary. Aliases for the same IP address can be added in different commands, if required.

Typically, static aliases are needed to access endpoints that do not belong to a zone (that is, they are not registered with any gatekeeper), or whose gatekeeper is inaccessible for some reason.

Examples

The following example creates a static terminal alias in the local zone:

```
zone local gk.zone1.com zone1.com
alias static 191.7.8.5 gkid gk.zone1.com terminal e164 14085551212 h323id bobs_terminal
```

alt-dial

To configure an alternate dial-out string for dial peers on the Cisco MC3810, use the **alt-dial** command in dial-peer configuration mode. To delete the alternate dial-out string, use the **no** form of this command.

alt-dial *string*

no alt-dial *string*

Syntax Description

string	The alternate dial-out string.

Defaults

No alternate dial-out string is configured.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Cisco MC3810 POTS, VoFR, VoATM, and VoHDLC dial peers.

The **alt-dial** command is used for the on-net-to-off-net alternative dialing function. The string replaces the destination-pattern string for dialing out.

Examples

The following example configures an alternate dial-out string of 9,5559871:

```
alt-dial 9,5559871
```

answer-address

To specify the full E.164 telephone number to be used to identify the dial peer of an incoming call, use the **answer-address** command in dial-peer configuration mode. To disable the configured telephone number, use the **no** form of this command.

answer-address [+]*string*[**T**]

no answer-address

Syntax Description

+	(Optional) Character indicating an E.164 standard number.
string	Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are: • Digits 0 through 9, letters A through D, pound sign (#), and asterisk (*), which represent specific digits that can be entered. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit.
T	(Optional) Control character indicating that the **answer-address** value is a variable length dial-string.

Defaults

The default value is enabled with a null string.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command is applicable to both Cisco 3600 series VoIP and POTS dial peers.

Use the **answer-address** command to identify the origin (or dial peer) of incoming calls from the IP network. Cisco IOS software identifies the dial peers of a call in one of two ways: either by identifying the interface through which the call is received or through the telephone number configured with the **answer-address** command. In the absence of a configured telephone number, the peer associated with the interface will be associated with the incoming call.

For calls coming in from a POTS interface, the **answer-address** command is not used to select an incoming dial peer. The incoming POTS dial peer is selected on the basis of the port configured for that dial peer.

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

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

Examples

The following example configures the E.164 telephone number, 555-9626, as the dial peer of an incoming call:

```
dial-peer voice 10 pots
 answer-address +5559626
```

Related Commands

Command	Description
destination-pattern	Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer.
port (dial peer)	Enables an interface on a PA-4R-DTR port adapter to operate as a concentrator port.
prefix	Specifies the prefix of the dialed digits for this dial peer.

application

To associate a specific interactive voice response (IVR) application with a POTS dial peer, use the **application** command in dial-peer configuration mode. To discontinue this association, use the **no** form of this command.

application *name*

no application *name*

Syntax Description

name	Indicates the name of the predefined IVR application. Incoming calls using this POTS dial peer will be handed off to this application.

Defaults

No default behavior or values.

Command Modes

Dial-peer configuration mode

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Usage Guidelines

Use this command when configuring interactive voice response (IVR) or any of the IVR-related features to associate a predefined session application with an incoming POTS dial peer. Calls using this incoming POTS dial peer will be handed to the predefined specified session application.

Examples

This following example shows how to define an application and how to apply it to an incoming POTS dial peer:

```
call application voice c4 tftp://santa/username/clid_4digits_npw_3.tcl
!
dial-peer voice 100 pots
 application c5
 answer-address 1234
 destination-pattern 100
 port 1/0/0
```

Related Commands

Command	Description
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice load	Reloads this designated TCL script.
call application voice pin-length	Defines the number of characters in the PIN for the application and passes that information to the application.

Command	Description
call application voice redirect-number	Defines the telephone number to which a call will be redirected—for example, the operator telephone number of the service provider—for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-length	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines the number of seconds a user is warned before their allowed calling time runs out for the designated application.

arq reject-unknown-prefix

To enable the gatekeeper to reject Admission Requests (ARQs) for zone prefixes that are not configured, use the **arq reject-unknown-prefix** command in gatekeeper configuration mode. To reenable the gatekeeper to accept and process all incoming ARQs, use the **no** form of this command.

arq reject-unknown-prefix

no arq reject-unknown-prefix

Syntax Description

This command has no arguments or keywords.

Defaults

The gatekeeper accepts and processes all incoming ARQs.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(6)Q, 11.3(7)NA, and 12.0(3)T	This command was introduced.

Usage Guidelines

Use the **arq reject-unknown-prefix** command to configure the gatekeeper to reject any incoming ARQs for a destination E.164 address that does not match any of the configured zone prefixes.

When an endpoint or gateway initiates an H.323 call, it sends an ARQ to its gatekeeper. The gatekeeper uses the configured list of zone prefixes to determine where to direct the call. If the called address does not match any of the known zone prefixes, the gatekeeper attempts to *hairpin* the call out through a local gateway. If you do not want your gateway to do this, then use the **arq reject-unknown-prefix** command. (*hairpin* is a term used in telephony that means to send a call back in the direction that it came from. For example, if a call cannot be routed over IP to a gateway that is closer to the target phone, the call is typically sent back out the local zone, back the way it came from.)

This command is typically used to either restrict local gateway calls to a known set of prefixes or deliberately fail such calls so that an alternate choice on a gateway's rotary dial-peer is selected.

Examples

Consider a gatekeeper configured as follows:

```
zone local gk408 cisco.com
zone remote gk415 cisco.com 172.21.139.91
zone prefix gk408 1408.......
zone prefix gk415 1415.......
```

In this example configuration, the gatekeeper manages a zone containing gateways to the 408 area code, and it knows about a peer gatekeeper with gateways to the 415 area code. Using the **zone prefix** command, the gatekeeper is then configured with the appropriate prefixes so that calls to those area codes hop off in the optimal zone.

If the **arq request-unknown-prefix** command is not configured, the gatekeeper handles calls in the following way:

- A call to the 408 area code is routed out through a local gateway.
- A call to the 415 area code is routed to the gk415 zone where it hops off on a local gateway there.
- A call to the 212 area code is routed to a local gateway in the gk408 zone.

If the **arq reject-unknown-prefix** command is configured, the gatekeeper handles calls in the following way:

- A call to the 408 area code is routed out through a local gateway.
- A call to the 415 area code is routed to the gk415 zone where it hops off on a local gateway there.
- A call to the 212 area code is rejected, because the destination address does not match any configured prefixes.

atm compression

To specify the software compression mode on the Cisco MC3810, use the **atm compression** command in interface configuration mode. To remove the compression mode setting, use the **no** form of this command.

atm compression {**per-packet** | **per-interface** | **per-vc**}

no atm compression {**per-packet** | **per-interface** | **per-vc**}

Syntax Description

per-packet	Specifies packet-by-packet compression mode (no history). This is the default.
per-interface	Specifies one context per interface (with history).
per-vc	Specifies one context for every virtual circuit (with history).

Defaults

per-packet

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

Examples

The following example configures per-packet ATM compression:

```
interface atm0
 atm compression per-packet
```

atm scramble-enable

To enable scrambling on E1 links, use the **atm scramble-enable** command in interface configuration mode. To disable scrambling, use the **no** form of this command.

atm scramble-enable

no atm scramble-enable

Syntax Description

This command has no arguments or keywords.

Defaults

By default, payload scrambling is off.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for ATM interface configuration on the Cisco MC3810.

Usage Guidelines

Enable scrambling on E1 links only. On T1 links, the default B8ZS line encoding normally ensures sufficient reliability. Scrambling improves data reliability on E1 links by randomizing the ATM cell payload frames to avoid continuous nonvariable bit patterns and improve the efficiency of the ATM cell delineation algorithms.

The scrambling setting must match that of the far end.

Examples

On a Cisco MC3810, the following example shows how to set the ATM0 E1 link to scramble payload:

```
interface atm0
 atm scramble-enable
```

atm video aesa

To set the unique ATM end-station address (AESA) for an ATM video interface that is using switched virtual circuit (SVC) mode, use the **atm video aesa** command in ATM interface configuration mode. To remove any configured address for the interface, use the **no** form of this command.

atm video aesa [default | *esi-address*]

no atm video aesa

Syntax Description

default	(Optional) Automatically creates an NSAP address for the interface, based on a prefix from the ATM switch (26 hexadecimal characters), the MAC address (12 hexadecimal characters) as the ESI (end station identifier), and a selector byte (two hexadecimal characters).
esi-address	(Optional) Defines the 12 hexadecimal characters used as the end-station identifier (ESI). The ATM switch provides the prefix (26 hexadecimal characters), and the video selector byte provides the remaining two hexadecimal characters.

Defaults

The **default** keyword is the default.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for ATM interface configuration on the Cisco MC3810.

Usage Guidelines

You cannot specify the ATM interface NSAP address in its entirety. The system creates either all of the address or part of it, depending on how you use this command.

Examples

On a Cisco MC3810, the following example shows the ATM interface NSAP address set automatically:

```
interface atm0
 atm video aesa default
```

On a Cisco MC3810, the following example shows the ATM interface NSAP address set to a specific ESI value:

```
interface atm0/1
 atm video aesa 444444444444
```

Related Commands

Command	Description
show atm video-voice address	Displays the NSAP address for the ATM interface.

audio-prompt load

To initiate loading the selected audio file (.au) (the file that contains the announcement prompt for the caller) from Flash memory into RAM, use the **audio-prompt load** command in privileged EXEC mode.

audio-prompt load *name*

Syntax Description

name	Indicates the location of the audio file that you want to be loaded from memory, Flash memory, or an FTP server. Presently, with Cisco IOS Release 11.3(6)NA2, the URL pointer refers to the directory where Flash memory is stored.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Usage Guidelines

The first time the IVR application plays a prompt, it reads it from the URL (or the specified location for the .au file, such as Flash or TFP) into RAM. Then it plays the script from RAM. An example of the sequence of events are:

- When the first caller is asked to enter their account and PINs, the enter_account.au and enter_pin.au files are loaded into RAM from Flash memory.
- When the next call comes in, these prompts are played from the RAM copy.
- If all callers enter valid account numbers and PINs, then the auth_failed.au file is not loaded from Flash memory into RAM memory.

The router will only load the audio file when the script initially plays that prompt after the router restarts. If the audio file is changed, you must run this EXEC command to reread the file. This will generate an error message if the file is not accessible, or if there is a format error.

Examples

The following example shows how to load the enter_pin.au audio file from Flash memory into RAM

```
audio-prompt load flash:enter_pin.au
```

auto-cut-through

To enable the Cisco MC3810 to complete a call when a PBX does not provide an M-lead response, use the **auto-cut-through** command in voice-port configuration mode. To disable the auto-cut-through operation, use the **no** form of this command.

auto-cut-through

no auto-cut-through

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

The **auto-cut-through** command applies to E&M voice ports only on the Cisco MC3810.

Examples

The following example enables the Cisco MC3810 to complete a call when a PBX does not provide an M-lead response:

```
voice-port 1/1
 auto-cut-through
```

busyout forced

To force the voice port into busyout state, use the **busyout forced** command in voice-port configuration mode. To remove the voice port from busyout state, use the **no** form of this command.

busyout forced

no busyout forced

Syntax Description

This command has no arguments or keywords.

Defaults

The voice-port is not in busyout state.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

When you force a voice port into busyout state, you must enter the **no busyout forced** command to remove the busyout state from the voice port.

Examples

The following example configures the voice port on the Cisco MC3810 into forced busyout state:

```
voice-port 1/1
 busyout forced
```

Related Commands

Command	Description
busyout-monitor interface interface	Places a voice port on the Cisco MC3810 multiservice concentrator into the busyout monitor state.
busyout-seize	Changes the busyout seize procedure fro a voice port on the Cisco MC3810 multiservice concentrator.
show voice busyout	Displays information about the voice busyout state on the Cisco MC3810 multiservice concentrator.
voice-port busyout	Places all voice ports associated with a serial or ATM interface into a busyout state.

busyout-monitor interface

To place a voice port into busyout monitor state, enter the **busyout-monitor interface** command in voice-port configuration mode. To remove the busyout monitor state on the voice port, use the **no** form of this command.

busyout-monitor interface *interface number*

no busyout-monitor interface *interface number*

Syntax Description

interface	The name of the associated interface or subinterface that will be monitored to trigger a voice-port busyout, for example **serial**, **atm**, or **ethernet**.
number	The slot and port position of the interface or subinterface, for example, *0/1*, *1/1.0*, and so on.

Defaults

The voice port is not in busyout monitor state.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(3)T	This command was introduced for the Cisco MC3810.
12.0(5)XK and 12.0(7)T	The command was modified for the Cisco 2600 and 3600 series.
12.0(5)XE and 12.0(7)T	The command was modified for the Cisco 7200 series routers.

Usage Guidelines

When you place a voice port in busyout monitor state, the voice port monitors the specified interface and enters the busyout state when the interface is down. This forces rerouting of calls when an interface is down.

If you specify more than one monitored interface for a voice port, all the monitored interfaces must be down in order to trigger busyout on the voice port.

The command monitors only the up or down status of an interface—not end-to-end TCP/IP connectivity.

When an interface is operational, a busied-out voice port returns to its normal state.

This feature can monitor LAN, WAN, and virtual interfaces, as well as subinterfaces.

Examples

The following example configures the voice port to monitor two serial interfaces and an Ethernet interface. When all these interfaces are down, the voice port is busied out. When at least one interface is operating, the voice port is put back into a normal state.

```
voice-port 3/0:0
 busyout monitor interface Ethernet0/0
 busyout monitor interface Serial1/0
 busyout monitor interface Serial2/0
```

Related Commands

Command	Description
busyout forced	Forces a voice port on the Cisco MC3810 multiservice concentrator into the busyout state.
busyout-seize	Changes the busyout seize procedure fro a voice port on the Cisco MC3810 multiservice concentrator.
show voice busyout	Displays information about the voice busyout state on the Cisco MC3810 multiservice concentrator.
voice-port busyout	Places all voice ports associated with a serial or ATM interface into a busyout state.

busyout-seize

To change the busyout seize procedure for a voice port, use the **busyout-seize** command in voice-port configuration mode. To restore the default busyout seize state on the voice port, use the **no** form of this command.

busyout-seize {**ignore** | **repeat**}

no busyout-seize {**ignore** | **repeat**}

Syntax Description

ignore	On busyout, leaves the loop open and ignores the incoming signal.
repeat	On busyout, seizes the far end and ignores all incoming signals until the far end release. Remove the seize signal, wait for one second before starting to seize the far end again.

Defaults

On busyout, the loop is closed and remains in the busyout state.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command is only supported on the Cisco MC3810.

Examples

The following example configures the busyout seize to the ignore state:

```
voice-port 1/1
 busyout-seize ignore
```

Related Commands

Command	Description
busyout forced	Forces a voice port on the Cisco MC3810 multiservice concentrator into the busyout state.
busyout-monitor interface interface	Places a voice port on the Cisco MC3810 multiservice concentrator into the busyout monitor state.
show voice busyout	Displays information about the voice busyout state on the Cisco MC3810 multiservice concentrator.
voice-port busyout	Places all voice ports associated with a serial or ATM interface into a busyout state.

cable arp

To activate cable Address Resolution Protocol (ARP), use the **cable arp** command in cable interface configuration mode. To disable cable ARP, use the **no** form of this command.

cable arp

no cable arp

Syntax Description

This command has no arguments or keywords.

Defaults

ARP enabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

ARP is an Internet protocol used to map IP addresses to MAC addresses on computers and other equipment installed in a network. You need to activate ARP requests so the Cisco uBR7200 series can perform IP address resolution on the downstream path.

Examples

The following example activates cable ARP requests for port 0 on the cable modem card installed in slot 6 of the Cisco uBR7200 series:

```
interface cable 6/0
 cable arp
```

Related Commands

Command	Description
cable proxy-arp	Activates cable proxy ARP on the cable interface.

cable channel-change

To move a cable modem from its current upstream channel to another upstream channel, use the **cable channel-change** command in cable interface configuration mode. To disable this feature, use the **no** form of this command.

cable channel-change *sid channel*

no cable channel-change *sid channel*

Syntax Description

sid	Service identifier (SID) of the cable modem. Valid values are from 1 to 8191.
channel	User-defined or user-selected. Valid values are from 0 to 6 depending on which cable modem line card is installed in the Cisco uBR7200 series.

Defaults

0

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

Moving a cable modem to a new channel can improve performance, increase bandwidth availability, or troubleshoot a cable modem. The SID identifies the particular cable modem you wish to move.

Examples

The following example moves the cable modem identified with SID 50 to channel 4:

```
interface cable 6/0
 cable channel-change 50 4
```

cable dhcp-giaddr

To modify the GIADDR field of DHCPDISCOVER and DHCPREQUEST packets with a Relay IP address before they are forwarded to the DHCP server, use the **cable dhcp-giaddr** command in cable interface configuration mode. To set the GIADDR field to its default, use the **no** form of this command.

cable dhcp-giaddr [**policy** | **primary**]

no cable dhcp-giaddr

Syntax Description

policy	(Optional) Selects the control policy, so the primary address is used for cable modems and the secondary addresses are used for hosts.
primary	(Optional) Always selects the primary address to be used for giaddr.

Defaults

No control of giaddr from the cable code.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

This command allows you to configure the cable modem subnet as the primary address and the host subnet as the secondary address.

Examples

The following example sets the primary address to be used always for giaddr.

```
interface cable 6/0
 cable dhcp-giaddr primary
```

cable downstream annex

To set the MPEG framing format for a downstream port on a cable modem card to either Annex A (Europe) or Annex B (North America), use the **cable downstream annex** command in cable interface configuration mode.

cable downstream annex {**A** | **B**}

Syntax Description

A	Annex A. The downstream is compatible with the European MPEG framing format specified in ITU-TJ.83 Annex A. This option is not supported in Cisco IOS Release 12.1.
B	Default. The downstream is compatible with the North American MPEG framing format specified in ITU-TJ.83 Annex B.

Defaults

Annex B

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

The MPEG framing format must be compatible with the downstream symbol rate you set. Annex B is the North America standard and Annex A is the European standard. You should review your local standards and specifications for downstream MPEG framing to determine which format you should use.

Note The cable modem card downstream ports and the cable modems on the HFC network connected through these ports must be set to the same MPEG framing format.

Caution In Cisco IOS Release 12.1, only Annex B MPEG framing format is supported.

Examples

The following example sets the MPEG framing format to Annex A:

```
interface cable 6/0
 cable downstream annex B
```

cable downstream channel-id

To configure the downstream channel ID, use the **cable downstream channel-id** command in cable interface configuration mode. To set the downstream channel ID to its default value, use the **no** form of this command.

cable downstream channel-id *id*

no cable downstream channel-id

Syntax Description

id	Specifies a downstream channel ID. Valid values are from 1 to 255.

Defaults

The unit number of the downstream device.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

Use this command to make sure each downstream channel has a unique ID when there are multiple Cisco uBR7200 series routers acting as CMTSes at a headend facility.

Cisco IOS assigns the default ID number of each downstream channel in the order in which devices connected to the downstream channels appear to the CMTS. The downstream channel connected to the first device that appears to the CMTS is configured with a default ID of 1, the downstream channel connected to the second device that appears is configured with an ID of 2, and so on. By assigning default values in this manner, a single CMTS guarantees unique channel IDs. However, this scheme does not guarantee unique channel IDs when more than one CMTS exists on a network.

Examples

The following example configures the downstream channel on the cable modem card in slot 6 of a Cisco uBR7200 series router with a channel ID of 44:

```
interface cable 6/0
 cable downstream channel-id 44
```

The following example restores the downstream channel id configuration to the default configuration:

```
interface cable 6/0
 cable downstream channel-id
```

cable downstream frequency

To set the fixed center frequency for downstream radio frequency carrier in hertz (Hz), use the **cable downstream frequency** command in cable interface configuration mode. To set no fixed center frequency, use the **no** form of this command.

cable downstream frequency *down-freq-hz*

no cable downstream frequency

Syntax Description

down-freq-hz	The known center frequency of the downstream carrier in Hz. The valid range is 54,000,000 to 1,020,000,000 Hz.

Defaults

Disabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

The downstream frequency of your RF output must be set to match the expected input frequency of your upconverter. To do this, you enter the fixed center frequency of the downstream channel for the downstream port. (You can also select a default which does not set a specific fixed value.) The valid range for a fixed center frequency is 54,000,000 to 1,020,000,000 Hz. The center frequency is also used to configure an IF-to-RF upconverter that must be installed in your downstream path.

The digital carrier frequency is specified to be the center of a 6.0 MHz channel. For example, EIA channel 95 spans 90.000 to 96.000 MHz. The center frequency is 93.000 MHz, which is the digital carrier frequency that should be configured as the downstream frequency. The typical range for current CATV headends is 88,000,000 to 860,000,000 Hz.

Note This command currently has no effect on external upconverters; it is informational only.

Examples

The following example sets the downstream center frequency:

```
interface cable 6/0
 cable downstream frequency 96000000
```

cable downstream if-output

To activate a downstream port on a cable modem card for digital data transmissions over the HFC network, use the **cable downstream if-output** command in cable interface configuration mode. To disable the 44-MHz intermediate frequency (IF) carrier, use the **no** form of this command.

cable downstream if-output

no cable downstream if-output

Syntax Description

This command has no arguments or keywords.

Defaults

Downstream carrier enabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Examples

The following example enables the downstream port 0 on the cable modem card installed in slot 6 of a Cisco uBR7200 series:

```
interface cable 6/0
 cable downstream if-output
```

cable downstream interleave-depth

To set the downstream interleave depth, use the **cable downstream interleave-depth** command in cable interface configuration mode. To restore the default setting, use the **no** form of this command.

cable downstream interleave-depth {**8** | **16** | **32** | **64** | **128**}

no cable downstream interleave-depth

Syntax Description

8 | **16** | **32** | **64** | **128**	Indicates the downstream interleave depth in milliseconds.

Defaults

32

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

This command sets the minimum latency of the system. A higher interleave depth provides more protection from bursts of noise on the HFC network; however, higher depth also increases downstream latency. Table 3 below shows interleave characteristics and their relation to each other.

Table 3 Interleave Characteristics and Relationships

I (Number of Taps)	J (Increment)	Burst Protection 64 QAM/256 QAM	Latency 64 QAM/256 QAM
8	16	5.9 milliseconds/4.1 milliseconds	0.22 ms/0.15 ms
16	8	12 milliseconds/8.2 milliseconds	0.48 ms/0.33 ms
32	4	24 milliseconds/16 milliseconds	0.98 ms/0.68 ms
64	2	47 milliseconds/33 milliseconds	2.0 ms/1.4 ms
128	1	95 milliseconds/66 milliseconds	4.0 ms/2.8 ms

Examples

The following example configures the downstream interleave depth to 128 milliseconds:

```
interface cable 6/0
 cable downstream interleave-depth 128
```

cable downstream modulation

To set the modulation rate for a downstream port on a cable modem card, use the **cable downstream modulation** command in cable interface configuration mode.

cable downstream modulation {**64qam** | **256qam**}

Syntax Description

64qam	Modulation rate is 6 bits per downstream symbol.
256qam	Modulation rate is 8 bits per downstream symbol.

Defaults

64qam

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

Downstream modulation defines the speed in symbols per second at which data travels downstream to the subscriber's cable modem. A symbol is the basic unit of modulation. QPSK encodes 2 bits per symbol, 16-QAM encodes 4 bits per symbol, 64-QAM encodes 6 bits per symbol, and 256-QAM encodes 8 bits per symbol.

Note Setting a downstream modulation rate of 256-QAM requires approximately a 6 dB higher signal-to-noise ratio (SNR) than 64-QAM at the subscriber's cable modem. If your network is marginal or unreliable at 256-QAM, use the 64-QAM format instead.

Examples

The following example sets the downstream modulation to 256-QAM:

```
interface cable 6/0
 cable downstream modulation 256qam
```

cable downstream rate-limit

To enable DOCSIS rate limiting on downstream traffic, use the **cable downstream rate-limit** command in cable interface configuration mode. To disable DOCSIS rate limiting on downstream traffic, use the **no** form of this command.

cable downstream rate-limit [**token-bucket** [[**shaping** [**granularity** *msec* | **max-delay** *msec*]] | **weighted-discard**] [*exp-weight*]

no cable downstream rate-limit

Syntax Description

token-bucket	(Optional) Specifies the token bucket filter algorithm.
shaping	(Optional) Enables rate limiting on the downstream port using the token bucket policing algorithm with default traffic shaping parameters.
granularity *msec*	(Optional) Specifies traffic shaping granularity in milliseconds. Valid values are 1, 2, 4, 8, or 16 milliseconds.
max-delay *msec*	(Optional) Specifies the maximum traffic shaping buffering delay in milliseconds. Valid values are 128, 256, 512, or 1028 milliseconds.
weighted-discard	(Optional) Specifies the weighted discard algorithm.
exp-weight	(Optional) Specifies the weight for the exponential moving average of loss rate. Valid values are from 1 to 4.

Defaults

cable downstream rate-limit, which enforces strict DOCSIS-complaint rate limiting.

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3(6) NA	This command was introduced.
12.0(4)XI	The **shaping** keyword was added.

Usage Guidelines

When you enter this command without an option, it enables strict DOCSIS-compliant rate limiting, which sets the burst rate to the interface speed.

Examples

The following example applies the token bucket filter algorithm:

```
interface cable 6/0
 cable downstream rate-limit token-bucket
```

Related Commands

Command	Description
cable upstream rate-limit	Sets DOCSIS rate limiting for an upstream port on a cable modem card.

cable flap-list aging

To specify the number of days to keep a cable modem in the flap-list table before aging it out of the table, use the **cable flap-list aging** command in global configuration mode. To disable this feature, use the **no** form of this command.

cable flap-list aging *number-of-days*

no cable flap-list aging

Syntax Description

number-of-days	Specifies how many days of cable modem performance is retained in the flap list. Valid values are from 1 to 60 days.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Flapping refers to the rapid disconnecting and reconnecting of a cable modem that is having problems holding its connection to the CMTS. A flap list is a table maintained by the Cisco uBR7200 series for every modem (active or not) that is having communication difficulties. The flap list contains modem MAC addresses and logs the time of the most recent activity. You can configure the size and entry thresholds for the flap list.

Examples

The following example specifies that the flap-list table retain two days of performance for this cable modem:

```
cable flap-list aging 2
```

Related Commands

Command	Description
cable flap-list insertion-time	Sets the insertion time interval that determines whether a cable modem is placed in the flap list.
cable flap-list power-adjust threshold	Specifies the power-adjust threshold for recording a cable modem flap-list event.
cable flap-list size	Specifies the maximum number of cable modems reported in the flap-list table.
clear cable flap-list	Resets the Cisco uBR7200 series flap-list table.

cable flap-list insertion-time

To set the cable flap-list insertion time interval, use the **cable flap-list insertion-time** command in global configuration mode. To disable insertion time, use the **no** form of this command.

cable flap-list insertion-time *seconds*

no cable flap-list insertion-time

Syntax Description

seconds	Insertion time interval in seconds. Valid values are from 60 to 86400 seconds.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

This command controls the operation of a flapping modem detector. When a cable modem makes insertion requests more frequently than the period of time defined by this command, the cable modem is placed in the flap list.

Examples

The following example sets the insertion time interval to 62 seconds:

```
cable flap-list insertion-time 62
```

Related Commands

Command	Description
cable flap-list aging	Specifies the number of days a cable modem remains in the flap-list table before being aged out of the table.
cable flap-list power-adjust threshold	Specifies the power-adjust threshold for recording a cable modem flap-list event.
cable flap-list size	Specifies the maximum number of cable modems reported in the flap-list table.
clear cable flap-list	Resets the Cisco uBR7200 series flap-list table.

cable flap-list power-adjust threshold

To specify the power-adjust threshold for recording a flap-list event, use the **cable flap-list power-adjust threshold** command in global configuration mode. To disable power-adjust thresholds, use the **no** form of this command.

cable flap-list power-adjust threshold *dBmV*

no cable flap-list power-adjust threshold

Syntax Description

dBmV	Specifies the minimum power adjustment in decibels per millivolt that will result in a flap-list event. Valid values are from 1 to 10 dBmV.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

This command controls the operation of a flapping modem detector. When the power adjustment of a cable modem exceeds the configured threshold value, the modem is placed in the flap list.

Note A power adjustment threshold of less than 2 dBmV might cause excessive flap list event recording. Cisco recommends setting this threshold value to 3 dBmV or higher.

Examples

The following example shows the power-adjust threshold being set to 5 dBmV:

```
cable flap-list power-adjust threshold 5
```

Related Commands

Command	Description
cable flap-list aging	Specifies the number of days a cable modem remains in the flap-list table before being aged out of the table.
cable flap-list insertion-time	Sets the insertion time interval that determines whether a cable modem is placed in the flap list.
cable flap-list size	Specifies the maximum number of cable modems reported in the flap-list table.
clear cable flap-list	Resets the Cisco uBR7200 series flap-list table.

cable flap-list size

To specify the maximum number of cable modems that can be listed in the flap-list table, use the **cable flap-list size** command in global configuration mode. To specify the default flap-list table size, use the **no** form of this command.

cable flap-list size *number*

no cable flap-list size

Syntax Description

number	Specifies the maximum number of cable modems that will report flap performance to the flap-list table. Valid values are from 1 to 8192.

Defaults

8192

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following example limits the flap-list table size to no more than 200 modems:

```
cable flap-list size 200
```

Related Commands

Command	Description
cable flap-list aging	Specifies the number of days a cable modem remains in the flap-list table before being aged out of the table.
cable flap-list insertion-time	Sets the insertion time interval that determines whether a cable modem is placed in the flap list.
cable flap-list power-adjust threshold	Specifies the power-adjust threshold for recording a cable modem flap-list event.
clear cable flap-list	Resets the Cisco uBR7200 series flap-list table.

cable helper-address

To specify a destination IP address for User Datagram Protocol (UDP) broadcast (DHCP) packets, use the **cable helper-address** command in cable interface configuration mode. To disable this feature, use the **no** form of this command.

cable helper-address *IP-address* {**cable-modem** | **host**}

no cable helper-address *IP-address* {**cable-modem** | **host**}

Syntax Description

IP-address	The IP address of a DHCP server to which UDP broadcast packets will be sent.
cable-modem	Specifies that only cable modem UDP broadcasts are forwarded.
host	Specifies that only host UDP broadcasts are forwarded.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

If you specify a secondary IP address, the GIADDR field in the DHCP requests will be sent to the primary address for DHCP requests received from cable modems, and to the secondary IP address for DHCP requests received from hosts.

Examples

The following example forwards UDP broadcasts from cable modems to the DHCP server at 172.23.66.44:

```
interface cable 6/0
 cable helper-address 172.23.66.44 cable-modem
```

The following example forwards UDP broadcasts from hosts to the DHCP server at 172.23.66.44:

```
interface cable 6/0
 cable helper-address 172.23.66.44 host
```

cable insertion-interval

To limit the amount of time that a cable modem can request an upstream frequency for the first time from the Cisco uBR7200 series, use the **cable insertion-interval** command in cable interface configuration mode. To configure the automatic setting and ignore any minimum or maximum time settings, use the **no** form of this command.

cable insertion-interval [**automatic**] [*min* | *max*]

no cable insertion-interval

Syntax Description

automatic	(Optional) Causes the Cisco uBR7200 series MAC scheduler for each upstream cable modem to vary the initial ranging times available to new cable modems joining the network.
min	(Optional) Minimum time in milliseconds that the Cisco uBR7200 series is allowed to specify in MAP messages as the initial ranging time for cable modems. Valid values are from 25 to 200 milliseconds. Default is 50 milliseconds.
max	(Optional) Maximum time in milliseconds that the Cisco uBR7200 series is allowed to specify in MAP messages as the initial ranging time for cable modems. Valid values are from 500 to 2000 milliseconds. Default is 2000 milliseconds (2 seconds).

Defaults

Automatic

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use this command to specify the minimum and maximum ranging times that will appear in MAP messages sent by the Cisco uBR7200 series. MAP messages define the precise time intervals during which modems can send.

The default insertion interval setting (**automatic**) configures the Cisco uBR7200 series to automatically vary (between 50 milliseconds and 2 seconds) the initial ranging times available to new cable modems that attempt to join the network.

Use the **automatic** keyword with this command when you have to bring a large number of modems online quickly (for example, after a major power failure). Override the **automatic** keyword by specifying an insertion interval.

Examples

The following example specifies automatic insertion intervals that vary from 50 ms to 2000 ms:

```
interface cable 6/0
 cable insertion-interval automatic
```

The following example specifies automatic insertion intervals that vary from 500 ms to 2000 ms:

```
interface cable 6/0
 cable insertion-interval min 500
```

The following example specifies automatic insertion intervals that vary from 50 ms to 1000 ms:

```
interface cable 6/0
 cable insertion-interval max 1000
```

cable ip-multicast-echo

To enable IP multicast echo, use the **cable ip-multicast-echo** command in cable interface configuration mode. To disable IP multicast echo, use the **no** form of this command.

cable ip-multicast-echo

no cable ip-multicast-echo

Syntax Description

This command has no arguments or keywords.

Defaults

Enable

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Examples

The following example disables IP multicast echo:

```
interface cable 6/0
 no cable ip-multicast-echo
```

cable ip-broadcast-echo

To activate upstream IP broadcast echo so that the Cisco uBR7200 series can echo broadcast packets, use the **cable ip-broadcast-echo** command in cable interface configuration mode. To disable the upstream IP broadcast echo, use the **no** form of this command.

cable ip-broadcast-echo

no cable ip-broadcast-echo

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Examples

The following example activates IP broadcast echo:

```
interface cable 6/0
 cable ip-broadcast-echo
```

cable match address

To specify that IP multicast streams be encrypted, use the **cable match address** command in cable interface configuration mode. Use the **no** form of this command if you do not want to use encryption.

cable match address *access-list*

no cable match address

Syntax Description

access-list	Specifies that the IP multicast streams defined by the access list be encrypted. Access lists can be IP access list numbers or an IP access list name. Valid access list numbers are from 100 to 199.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Configure the access list using the **ip access-list** command. For information on this command, refer to the *Cisco IOS IP and IP Routing Command Reference* publication.

Examples

The following example specifies that the multicast stream defined by the access list named **reno** be encrypted:

```
interface cable 6/0
 cable match address reno
```

The following example specifies that the multicast stream defined by the access list number 102 be encrypted:

```
interface cable 6/0
 cable match address 102
```

Related Commands

Command	Description
ip access-list	Defines an IP access list by name.

cable modem change-frequency

To override the frequency used by a cable modem, use the **cable modem change-frequency** command in EXEC mode.

cable modem [*mac-addr* | *ip-addr*] **change-frequency** [**ds-frequency-hz**] [**us-channel-id**]

Syntax Description

mac-addr | *ip-addr*	(Optional) Specifies either the MAC address or the IP address of the cable modem whose frequency is to be changed.
ds-frequency-hz	(Optional) Specifies the downstream frequency for the cable modem (in Hertz).
us-channel-id	(Optional) Specifies the upstream channel ID.

Defaults

No default behavior or values.

Command Modes

EXEC

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following example changes the downstream frequency of the cable modem having IP address 172.172.172.12 to 57 MHz:

```
interface cable 6/0
 cable modem 172.172.172.12 change-freq 57000000
```

Related Commands

Command	Description
cable modem max-hosts	Specifies the maximum number of hosts supported by a specific cable modem.

cable modem max-hosts

To specify the maximum number of CPE devices (hosts) that can be supported by a specific cable modem, use the **cable modem max-hosts** command in cable interface configuration mode. To set the number of hosts to 0, use the **no** form of this command.

cable modem {*mac-addr* | *ip-addr*} **max-hosts** {*n* | **default**}

no cable modem {*mac-addr* | *ip-addr*} **max-hosts**

Syntax Description

mac-addr | *ip-addr*	Specifies either the MAC address or the IP address of the cable modem.
max-hosts {*n* | **default**}	Specifies either the maximum number of hosts supported by the cable modem (from 0 to 255), or specifies the default value of 0.

Defaults

0

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following example limits the cable modem having IP address 172.172.172.12 to a maximum of 40 attached CPE devices:

```
interface cable 6/0
 cable modem 172.172.172.12 max-hosts 40
```

Related Commands

Command	Description
cable modem change-frequency	Overrides the frequency used by a cable modem.

cable modulation-profile

To define the cable modulation profile, use the **cable modulation-profile** command in global configuration mode. To remove the specified modulation profile, use the **no** form of this command.

cable modulation-profile *profile iuc fec-tbytes fec-len burst-len guard-t mod scrambler seed diff pre-len last-cw uw-len*

no cable modulation-profile *profile iuc fec-tbytes fec-len burst-len guard-t mod scrambler seed diff pre-len last-cw uw-len*

Syntax Description

profile	Modulation profile number.
iuc	Interval usage code. Valid entries are: **initial**, **long**, **request**, **short**, or **station**.
fec-tbytes	The number of bytes that can be corrected per FEC code word. Valid values are from 0 to 10, where 0 means no FEC.
fec-len	FEC code word length. Valid values are from 16 to 253.
burst-len	Maximum burst length in minislots. Valid values are from 0 to 255, where 0 means no limit.
guard-t	Guard time in symbols. The time between successive bursts.
mod	Modulation. Valid entries are **16qam** and **qpsk**.
scrambler	Enable or disable scrambler. Valid entries are **scrambler** and **no-scrambler**.
seed	Scrambler seed in hexidecimal format. Valid values are from 0x0000 to 0x7FFF.
diff	Enable or disable differential encoding. Valid entries are **diff** and **no-diff**.
pre-len	Preamble length in bits. Valid values are from 2 to 128.
last-cw	Handling of FEC for last code word. Valid entries are **fixed** for fixed code word length and **shortened** for shortened last code word.
uw-len	Upstream unique word length. Enter **uw8** for 8-bit unique words or **uw16** for 16-bit unique code words.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

You can use the **no** form of this command to remove all modulation profiles except modulation profile 1. In the case of modulation profile 1, the **no** form of this command sets all of the parameters in a burst to default values.

Caution Changes to modulation profiles causes changes to the physical layer. Because changing physical layer characteristics affects router performance and function, this task should be reserved for expert users.

Examples

The following example defines the burst parameters for profile 2:

The request burst is defined to have 0 fec-tbytes, 16 KB fec-len, a burst-len of 1, a guard time of 8, a mod value of qpsk, scrambler enabled with a seed value of 152, differential encoding disabled, a preamble length of 64 bits, a fixed code word length, and 8-bit unique words for upstream unique word length. The remaining initial, station, short, and long bursts are defined in similar fashion for profile 2.

```
cable modulation-profile 2 request 0 16 1 8 qpsk scrambler 152 no-diff 64 fixed uw8
cable modulation-profile 2 initial 5 34 0 48 qpsk scrambler 152 no-diff 128 fixed uw16
cable modulation-profile 2 station 5 34 0 48 qpsk scrambler 152 no-diff 128 fixed uw16
cable modulation-profile 2 short 6 75 6 8 16qam scrambler 152 no-diff 144 fixed uw8
cable modulation-profile 2 long 8 220 0 8 16qam scrambler 152 no-diff 160 fixed uw8
```

Note You have to create all of the bursts (request, initial, station, short and long) for this modulation profile to use the **modulation profile** command.

See the **show cable modulation-profile** command for a description of the output display fields.

Related Commands

Command	Description
cable upstream modulation-profile	Assigns a modulation profile to an interface on a cable router.
show cable modulation-profile	Displays modulation profile group information.

cable privacy

To enable privacy in the system, use the **cable privacy** command in cable interface configuration mode. To disable privacy, use the **no** form of this command.

cable privacy [**mandatory** | **authenticate-modem** | **authorize-multicast**]

no cable privacy

Syntax Description

mandatory	(Optional) Enforce Baseline Privacy for all modems.
authenticate-modem	(Optional) Use AAA protocols to authenticate all modems during BPI initialization.
authorize-multicast	(Optional) Use AAA protocols to authorize all multicast stream (IGMP) join requests.

Defaults

Mandatory

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

While the default for this command is to enable privacy, it is not mandatory.

Examples

The following examples all begin in cable interface configuration mode. The first example displays the options available with the **cable privacy** command:

```
cable privacy ?
 authenticate-modem   turn on BPI modem authentication
 authorize-multicast  turn on BPI multicast authorization
 kek                  KEK Key Parms
 mandatory            force privacy be mandatory
 tek                  TEK Key Parms
```

The following example forces Baseline Privacy to be used for all cable modems:

```
cable privacy mandatory
```

The following example turns on BPI modem authentication:

```
cable privacy authenticate-modem
```

The following example turns on BPI multicast authorization:

```
cable privacy authorize-multicast
```

Related Commands

Command	Description
ping cable-modem	Determines whether a specific cable modem is online.

cable privacy kek

To set key encryption keys (KEKs) grace-time and life-time values for baseline privacy on an HFC network, use the **cable privacy kek** command in global configuration mode. To restore the default values, use the **no** form of this command.

cable privacy kek {**grace-time** [*seconds*] | **life-time** [*seconds*]}

no cable privacy kek {**grace-time** | **life-time**}

Syntax Description

grace-time *seconds*	(Optional) Length of key encryption grace-time in seconds. Valid range is 300 to 1800 seconds.
life-time *seconds*	(Optional) Length of the key encryption life-time in seconds.Valid range is 86400 to 6048000.

Defaults

grace-time: 600 seconds

life-time: 604800 seconds

Command Modes

Global configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

Baseline privacy on an HFC network is configured with key encryption keys (KEKs) and traffic encryption keys (TEKs). The encryption is based on 40-bit or 56-bit data encryption standard (DES) encryption algorithms.

A KEK is assigned to a cable modem based on the cable modem service identifier (SID) and permits the cable modem to connect to the Cisco uBR7200 series when baseline privacy is activated. KEKs can be set to expire based on a grace-time or a life-time value.

The **grace-time** keyword is used to assign a temporary key to a cable modem to access the network. The **life-time** keyword is used to assign a more permanent key to a cable modem.

A cable modem that has a grace-time or life-time key assigned by the Cisco uBR7200 series will request a new key before the current one expires.

Examples

The following example sets the KEK privacy grace-time to 800 seconds:

```
cable privacy kek grace-time 800
```

The following example sets the KEK privacy life-time to 750000 seconds:

```
cable privacy kek life-time 750000
```

cable privacy tek

To set traffic encryption keys (TEKs) grace-time and life-time values for baseline privacy on an HFC network, use the **cable privacy tek** command in global configuration mode. To restore the default value, use the **no** form of this command.

cable privacy tek {**grace-time** [*seconds*] | **life-time** [*seconds*]}

no cable privacy {**tek grace-time** | **life-time**}

Syntax Description

grace-time *seconds*	(Optional) Length of traffic encryption grace-time in seconds. Valid range is 300 to 1800 seconds. Default is 600 seconds.
life-time *seconds*	(Optional) Length of the traffic encryption life-time in seconds.Valid range is 1800 to 6048000. Default is 43200 seconds.

Defaults

grace-time: 600 seconds

life-time: 43200 seconds

Command Modes

Global configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

Baseline privacy on an HFC network is configured with key encryption keys (KEKs) and traffic encryption keys (TEKs). The encryption is based on 40-bit or 56-bit data encryption standard (DES) encryption algorithms.

The TEK is assigned to a cable modem when its kek has been established. The TEK is used to encrypt data traffic between the cable modem and the Cisco uBR7200 series. TEKs can be set to expire based on a grace-time or a life-time value.

The **grace-time** keyword is used to assign a temporary key to a cable modem to access the network. The **life-time** keyword is used to assign a more permanent key to a cable modem.

A cable modem that has a grace-time or life-time key assigned by the Cisco uBR7200 series will request a new key before the current one expires.

Examples

The following example sets the traffic encryption key grace-time to 800 seconds:

```
cable privacy tek grace-time 800
```

The following example sets the traffic encryption key life-time to 800000 seconds:

```
cable privacy tek life-time 800000
```

cable proxy-arp

To activate cable proxy Address Resolution Protocol (ARP) on the cable interface, use the **cable proxy-arp** command in cable interface configuration mode. To disable this feature, use the **no** form of this command.

cable proxy-arp

no cable proxy-arp

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

Because the downstream and upstream are separate interfaces, cable modems cannot directly perform address resolution with other cable modems on the cable plant. This command allows cable modems to perform address resolution through a proxy.

Examples

The following example activates proxy ARP for host-to-host communications:

```
interface cable 6/0
 cable proxy-arp
```

cable qos permission

To specify permission for updating the quality of service (QoS) table, use the **cable qos permission** command in global configuration mode. To remove a previously enabled permission, use the **no** form of this command.

cable qos permission {**create-snmp** | **modems** | **update-snmp**}

no cable qos permission

Syntax Description

create-snmp	Permits creation of QoS table entries by Simple Network Management Protocol (SNMP).
modems	Permits creation of QoS table entries by modem registration requests.
update-snmp	Permits dynamic update of QoS table entries by SNMP.

Defaults

Enable by modem and SNMP.

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following example enables cable modems to request arbitrary QoS parameters:

```
cable qos permission modems
```

Related Commands

Command	Description
cable qos permission enforce	Overrides the provisioned QoS profile of the cable modem and enforces a specific CMTS-local QoS profile.
cable qos permission enforce	Configures a QoS profile.
show cable qos permission	Displays the status of permissions for changing QoS tables for a cable router.
show cable qos profile	Displays cable router QoS profiles.

cable qos permission enforce

To override the provisioned quality of service (QoS) profile of the cable modem and enforce a specific CMTS-local QoS profile, use the **cable qos permission enforce** command in global configuration mode. To allow cable modems to use the QoS profile they were configured to use, use the **no** form of this command.

cable qos permission enforce *index*

no cable qos permission enforce

Syntax Description

index	Specifies the number of the QoS profile to be enforced on all cable modems connecting to the CMTS. Valid values are from 1 to 255.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3(9)NA	This command was introduced.

Usage Guidelines

This command allows CMTS operators to enforce a profile on all connected cable modems to ensure that rate limiting is properly implemented.

If the QoS profile to be enforced does not exist at the CMTS during registration, the CMTS uses the QoS profile configured for the registering cable modem.

Examples

The following example shows how a cable modem with a QoS profile 4 created by the cable modem (*cm*) is reset to use QoS profile 225 enforced by the cable router (management):

```
CMTS01# show cable modem
 Interface    SID  Online     Timing Receive QoS IP address      MAC
 address
                   State      Offset Power
 Cable6/0/U0  1    online     2848    0.00    4   19.2.20.139 0010.7b6b.7215

CMTS01# show cable qos profile 4
 Service Prio Max       Guarantee Max        Max tx TOS  TOS   Create  B
 class        upstream  upstream  downstream burst  mask value by      priv
              bandwidth bandwidth bandwidth                            enab
 4       7    128000    64000     2048000    255    0x0  0x0   cm      no

CMTS01(config)# cable qos profile 225 max-upstream 256
CMTS01(config)# cable qos permission enforce 225
CMTS01# clear cable modem all reset
CMTS01# show cable modem
 Interface    SID  Online     Timing Receive QoS IP address      MACaddress
                   State      Offset Power
 Cable6/0/U0  1    offline    2848    0.25    2   19.2.20.139 0010.7b6b.7215

CMTS01# debug cable reg
 ....
 00:15:59: Finished parsing REG Request
 00:15:59: Overriding Provisioned QoS Parameters In REG-REQ
 ....

CMTS01# show cable modem
 Interface    SID  Online     Timing Receive QoS IP address      MACaddress
                   State      Offset Power
 Cable6/0/U0  1    online     2852    0.00    225 19.2.20.139  0010.7b6b.7215

CMTS01# show cable qos profile 225
 Service Prio Max       Guarantee Max        Max tx TOS  TOS   Create      B
 class        upstream  upstream  downstream burst  mask value by          priv
              bandwidth bandwidth bandwidth                                enab
 225     0    256000    0         0          0      0x0  0x0   management  no
```

Related Commands

Command	Description
cable qos permission	Specifies permission for updating the cable router QoS table.
cable qos profile	Configures a QoS profile.
show cable qos permission	Displays the status of permissions for changing QoS tables for a cable router.
show cable qos profile	Displays cable router QoS profiles.

cable qos profile

To configure a QoS profile, use the **cable qos profile** command in global configuration mode. To either set default values for profile group numbers 1 or 2, or to remove the QoS profile if no specific parameters remain, use the **no** form of this command.

cable qos profile {*groupnum* | *ip-precedence* | *guaranteed-upstream* | *max-burst* | *max-upstream* | *max-downstream* | *priority* | *tos-overwrite* | *value*}

no cable qos profile {*groupnum* | *ip-precedence* | *guaranteed-upstream* | *max-burst* | *max-downstream* | *priority* | *tos-overwrite* | *value*}

Syntax Description

groupnum	QoS profile group number. Qos profiles 1 and 2 are required by the system; they are preconfigured and can be modified but cannot be removed. QoS profile 1 is used during registration; QoS profile 2 is the default QoS profile.
ip-precedence	Sets the bits in the ToS byte that enable you to configure individual data rate limits on a per-modem basis. Valid values are from 0 to 7.
guaranteed-upstream	Guaranteed minimum upstream rate in kilobits per second. Valid values are from 0 to 100000 kbps. Default value is 0 (no reserved rate).
max-burst	Maximum upstream transmit burst size in minislots that the modem can send for any single transmit burst. Valid values are from 0 to 255 minislots. Default value is 0 (no limit).
max-upstream	Maximum upstream data rate in kilobits per second that a modem using this QoS profile will receive. Valid values are from 0 to 255 kbps. Default value is 0 (no upstream rate limit).
max-downstream	Maximum downstream data rate in kilobits per second that a modem using this QoS profile will receive. Valid values are from 0 to 255 kbps. Default value is 0 (no downstream rate limit).
priority	Relative priority number assigned to upstream traffic by this QoS profile. Valid values are from 0 to 7, with 7 being the highest priority. Default value is 0.
tos-overwrite	Overwrite the Type of Service (ToS) field in the IP datagrams received on the upstream before forwarding them downstream. This parameter sets the mask bits to a hexadecimal value to help the CMTS identify datagrams for QoS on the backbone.
value	The value substituted for the ToS value.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.
12.0(5)T	The *ip-precedence* argument was added and the range for the *max-downstream* argument was increased.

Examples

The following example configures QoS profile 4 with a guaranteed upstream rate of 8 kbps, maximum transmission burst of 16 minislots, maximum downstream rate of 128 kbps, with a priority of 4, cable baseline privacy set, and a tos-overwrite mask and value byte (in hex) of 0x2:

```
cable qos profile 4 guaranteed-upstream 8
cable qos profile 4 max-burst 16
cable qos profile 4 max-downstream 128
cable qos profile 4 priority 4
cable qos profile 4 tos-overwrite 0x2
```

Related Commands

Command	Description
cable qos permission	Specifies permission for updating the cable router QoS table.
show cable qos profile	Displays cable router QoS profiles.

cable relay-agent-option

To enable the system to insert the cable modem MAC address into a DHCP packet received from a cable modem or host and forward the packet to a DHCP server, use the **cable relay-agent-option** command in cable interface configuration mode. To disable MAC address insertion, use the **no** form of this command.

cable relay-agent-option

no cable relay-agent-option

Syntax Description

This command has no keywords or arguments.

Defaults

no cable relay-agent-option

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

This functionality enables a DHCP server to identify the user (cable modem) sending the request and initiate appropriate action based on this information.

Examples

The following example enables the insertion of DHCP relay agent information into DHCP packets:

```
interface cable 6/0
 cable relay-agent-option
```

cable shared-secret

To configure authentication and data privacy parameters, use the **cable shared-secret** command in cable interface configuration mode. To disable authentication during the cable modem registration phase, use the **no** form of this command.

cable shared-secret [**0** | **7**] *authentication-key*

no cable shared-secret

Syntax Description

0	(Optional) Specifies that an unencrypted message will follow.
7	(Optional) Specifies that an encrypted message will follow.
authentication-key	Text string is a shared secret string. When you enable the service password-encryption option, the password is stored in encrypted form. The text string is a 64-character authentication key.

Defaults

Disabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

Use this command to activate authentication so that all cable modems must return a known text string to register with the Cisco uBR7200 series for access to the HFC network.

Examples

The following example activates cable modem authentication using 3344912349988...sf as the shared secret key and indicating that an encrypted message follows:

```
interface cable 6/0
 cable shared-secret 7 3344912349988cisco@xapowenaspasdpuy230jhm...sf
```

cable source-verify

To turn on cable modem upstream verification, use the **cable source-verify** command in cable interface configuration mode. To disable verification, use the **no** form of this command.

cable source-verify [**dhcp**]

no cable source-verify dhcp

Syntax Description

dhcp	(Optional) Specifies that queries will be sent to verify unknown source IP addresses in upstream data packets.

Defaults

Disabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.
12.0(7)T	The **dhcp** keyword was added.

Usage Guidelines

The Cisco uBR7200 series sends DHCP LEASEQUERIES to verify unknown source IP addresses in upstream data packets. For maximum protection, turn on the DHCP relay-agent information option on the Cisco uBR7200 series when using this feature.

Examples

The following example turns on cable modem upstream verification and configures the Cisco uBR7200 series to send DHCP LEASEQUERIES to verify unknown source IP addresses in upstream data packets:

```
interface cable 6/0
 cable source-verify dhcp
```

Related Commands

Command	Description
cable relay-agent-option	Enables the system to insert the cable modem MAC address into a DHCP packet received from a cable modem or host and forwards the packet to a DHCP server.

cable spectrum-group

To create a spectrum group, use the **cable spectrum-group** command in global configuration mode. To disable this spectrum group, use the **no** form of this command.

cable spectrum-group *group-number* **type** {**blind** | **scheduled** | **daily periodic-sec** *seconds*}

no cable spectrum-group *group-number* **type** {**blind** | **scheduled** | **daily periodic-sec** *seconds*}

Syntax Description

group-number	Spectrum group number. Valid range is from 1 to 32.
blind	Creates a spectrum group that enables the upstream frequency and input power level to change whenever noise impairs upstream data traffic.
scheduled daily	Creates a spectrum group that enables the upstream frequency and power level to change at a set time during the day.
scheduled periodic-sec	Creates a spectrum group that enables the upstream frequency and power level to change at a specified interval in seconds.
seconds	Rate in seconds when upstream frequency and power level change.

Defaults

No spectrum group is defined.

Command Modes

Global configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

Upstream traffic may be affected by noise or other cable plant impairments. The spectrum manager monitors the upstream traffic. If station maintenance messages from cable modems are not received for approximately 2.5 minutes, the spectrum manager reassigns a different upstream frequency to the upstream channel.

Frequency agility is configured and activated using spectrum groups. A spectrum group is a table of frequencies that can be used by upstream ports to implement a frequency-hopping policy. There are two types of policies, blind and scheduled, with two corresponding types of spectrum groups.

- Blind—The spectrum manager automatically assigns a new upstream channel frequency when station maintenance (keepalive) messages fail for approximately 2.5 minutes. This represents a complete impairment of the upstream channel due to noise, plant, or equipment failure.
- Scheduled—The spectrum manager automatically assigns a new upstream frequency at set times during the day.

Note The cable interface will not operate until you either create and configure a spectrum group or set a fixed upstream frequency. From the interface configuration prompt, an interface is assigned membership in a spectrum group. From the interface point of view, the spectrum group also represents the set of upstreams connected to the same group of fiber nodes. This allows the spectrum manager to know if the upstream spectrum is shared.

A maximum of 32 spectrum groups can be configured in the system.

Examples

The following example creates three types of spectrum groups and sets the periodic rate to 48000 seconds:

```
cable spectrum-group 1 blind
cable spectrum-group 2 type scheduled daily
cable spectrum-group 3 type scheduled periodic-sec 48000
```

Related Commands

Command	Description
cable spectrum-group band	Configures a continuous frequency band setting for a cable spectrum group.
cable spectrum-group frequency	Configures a spectrum group to use a center frequency.

cable spectrum-group band

To configure a continuous frequency band setting for a spectrum group, use the **cable spectrum-group band** command in global configuration mode. To delete the band settings for a spectrum group, use the **no** form of this command.

cable spectrum-group *group-number* [**time** *day hh***:***mm***:***ss*] [**delete**] **band** *start-freq-hz end-freq-hz* [*power-level-dbmv*]

no cable spectrum-group *group-number*

Syntax Description

group-number	Spectrum group number. Valid values are from 1 to 32.
time *day hh:mm:ss*	(Optional) For scheduled spectrum groups, makes the frequency band setting available on the specified day at the specified time in hours (hh), minutes (mm), and seconds (ss).
delete	(Optional) Removes the frequency band setting from use at the specified time.
start-freq-hz	Lower boundary of the frequency band.
end-freq-hz	Upper boundary of the frequency band.
power-level-dbmv	(Optional) Nominal input power level in decibels per millivolt (dBmV). Valid values are from –10 to +10 dBmV. Some cable plants might want to change only the input power level and not frequency on a daily time schedule.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3 XA1	This command was introduced.

Usage Guidelines

This command specifies that a continuous frequency band setting be used as a unit of allocated spectrum within this spectrum group. Cable plants can choose to set up a daily schedule that changes the input power level and not the frequency.

Examples

The following example specifies that all the upstream ports for spectrum-group 4 share the same spectrum from 5000004 Hz to 40000000 Hz with a power level of 5 dBmV on Mondays at noon:

```
cable spectrum-group 4 time Monday 12:00:00 band 5000004 40000000 5
```

The following example deletes the frequency band created in the previous example:

```
cable spectrum-group 4 time Monday 12:00:00 delete band 5000004 40000000 5
```

Related Commands

Command	Description
cable spectrum-group	Creates a spectrum group of a specified type.
cable spectrum-group frequency	Configures a spectrum group to use a center frequency.

cable spectrum-group frequency

To configure a list of upstream frequencies and nominal power levels that each spectrum group can use when an upstream frequency change is necessary, use the **cable spectrum-group frequency** command in global configuration mode. To delete a spectrum group list, use the **no** form of this command.

cable spectrum-group *group-number* [**time** *day hh:mm:ss*] [**delete**] **frequency** *ctr-freq-hz* [*power-level-dbmv*]

no cable spectrum-group *group-number*

Syntax Description

group-number	Spectrum group number. Valid range is 1 to 32. Configuring a spectrum group with multiple entries of this type defines a list of frequencies which are available for use as upstream frequencies.
time *day hh***:***mm***:***ss*	(Optional) For scheduled spectrum groups, specifies the day and time of day that the frequency and input power level should change. Valid entries for the day argument are: Monday: **mon** Tuesday: **tue** Wednesday: **wed** Thursday: **thu** Friday: **fri** Saturday: **sat** Sunday: **sun** Valid entries for the *hh:* argument are 00 to 23. Valid entries for the *mm:* argument are 00 to 59. Valid entries for the *ss:* argument are 00 to 59.
delete	(Optional) Removes the frequency setting from use at the specified time.
ctr-freq-hz	Upstream frequency in Hz. Valid range is 5,000,000 to 42,000,000 Hz.
power-level-dbmv	(Optional) Nominal input power level. Valid range is –10 to +10 decibels per millivolt (dBmV). Some cable plants might want to change only the input power level and not the frequency on a daily time schedule.

Defaults

Operator must determine a value based on the spectrum allocation plan.

Command Modes

Global configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

After you create a spectrum group, you need to configure a list of upstream frequencies and nominal power levels that each spectrum group can use when an upstream frequency change is necessary. Each spectrum group should have its own list of upstream frequencies. Valid frequencies are 5,000,000 to 42,000,000 Hz; valid power levels are –10 dBmV to 10 dBmV. The power level value should only be entered if you want to change only the power level as part of spectrum management. The standard power level is 0 dBmV.

Examples

The following example creates spectrum group frequencies:

```
cable spectrum-group 1 frequency 6500000
cable spectrum-group 1 frequency 7000000
cable spectrum-group 2 frequency 750000 -5
cable spectrum-group 3 time 02:00:00 frequency 9000000
cable spectrum-group 3 time 12:00:00 frequency 9500000 -5
```

Related Commands

Command	Description
cable spectrum-group	Creates a spectrum group of a specified type.
cable spectrum-group band	Configures a continuous frequency band setting for a cable spectrum group.

cable spectrum-group hop period

To set the minimum frequency-hop interval, use the **cable spectrum-group hop period** command in global configuration mode. To delete the frequency hop interval for this spectrum group, use the **no** form of this command.

cable spectrum-group *groupnum* **hop period** *seconds*

no cable spectrum-group *groupnum* **hop period**

Syntax Description

groupnum	Spectrum group number. Valid values are from 1 to 32.
seconds	Specifies the frequency-hop time period in seconds. Valid values are from 1 to 3600 seconds. Default value is 300 seconds.

Defaults

300 seconds

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

The Cisco uBR7200 series router polls each cable modem at a default rate of once every 10 seconds. When ingress noise causes the loss of keepalive messages for a specified period of time, a new frequency is selected from the allocation table and a UCD update is performed.

If the destination channel is expected to be impaired, the minimum period between frequency hops can be reduced to a small value such as 10 seconds. This allows the frequency hop to continue more rapidly until a clear channel is found. If excessive frequency hop is a concern, the minimum period between hops can be increased.

Examples

The following example reduces the minimum frequency-hop interval to 60 seconds:

```
cable spectrum-group hop period 60
```

Related Commands

Command	Description
cable spectrum-group hop threshold	Specifies a hop threshold for a cable spectrum group.

cable spectrum-group hop threshold

To specify a frequency hop threshold for a spectrum group, use the **cable spectrum-group hop threshold** command in global configuration mode. To delete the hop threshold for this spectrum group, use the **no** form of this command.

cable spectrum-group *groupnum* **hop threshold** [*percent*]

no cable spectrum-group *groupnum* **hop threshold**

Syntax Description

groupnum	Spectrum group number. Valid values are from 1 to 32.
percent	(Optional) Specifies the frequency hop threshold as a percentage of cable modems going offline. Valid range is from 1 to 100 percent.

Defaults

100 percent

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

The Cisco uBR7200 series router polls each cable modem at a default rate of once every 10 seconds. When ingress noise causes loss of keepalive messages from a configurable percentage of all cable modems, resulting in those cable modems going offline, a new frequency is selected from the allocation table and a UCD update is performed.

Use this command to prevent a single failing cable modem from affecting service to other good cable modems.

Examples

The following example sets the threshold that triggers frequency hop to 20 percent of all cable modems for spectrum-group 4:

```
cable spectrum-group 4 hop threshold 20
```

Related Commands

Command	Description
cable spectrum-group hop period	Sets the minimum frequency-hop interval for a cable spectrum group.

cable spectrum-group shared

To specify that the upstream ports in a spectrum group share the same upstream frequency, use the **cable spectrum-group shared** command in global configuration mode. To delete this specification, use the **no** form of this command.

cable spectrum-group *groupnum* **shared**

no cable spectrum-group *groupnum* **shared**

Syntax Description

groupnum	Spectrum group number. Valid values are from 1 to 32.

Defaults

Upstream port frequency the same for all ports in the spectrum group.

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Because this command forces upstream ports to use the same spectrum, do not use this command for overlapping carriers.

Examples

The following example specifies that all the upstream ports for spectrum-group 4 share the same upstream frequency:

```
cable spectrum-group 4 shared
```

cable telco-return enable

To enable telephone return support, use the **cable telco-return enable** command in cable interface configuration mode. To disable telephone return support, use the **no** form of this command.

cable telco-return enable

no cable telco-return enable

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following example enables telephone return support:

```
interface cable 6/0
 cable telco-return enable
```

Related Commands

Command	Description
cable telco-return interval	Specifies the intervals for sending TCD enrollment messages and TSI messages in a cable-routed system.

cable telco-return interval

To specify the intervals for sending Telephony Channel Descriptor (TCD) enrollment messages and transmitting subscriber information (TSI) messages, use the **cable telco-return interval** command in cable interface configuration mode. To set the time interval to the default, use the **no** form of this command.

cable telco-return interval *seconds*

no cable telco-return interval

Syntax Description

seconds	Specifies the interval in seconds for sending TCD or TSI messages. Valid range is from 2 to 60 seconds. Default value is 2 seconds.

Defaults

2 seconds

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Usage Guidelines

Downstream traffic must be precluded by TCD messages to enable upstream telco return traffic. TCD messages contain information necessary for the telco return cable modem to access the headend/ISP network access server over the PSTN.

Use this command to specify how often TCD and TSI messages are sent.

Examples

The following example specifies the interval for sending TCD enrollment messages and transmitting subscriber information (TSI) messages as 40 seconds:

```
interface cable 6/0
cable telco-return interval 40
```

Related Commands

Command	Description
cable telco-return enable	Enables telephone return support in a cable-routed system.

cable telco-return spd dhcp-authenticate

To set the DHCP Authenticate parameter in Telephony Channel Descriptor (TCD) messages to TRUE (1), specifying the DHCP server that must be used, use the **cable telco-return spd dhcp-authenticate** command in cable interface configuration mode. To set the DHCP Authenticate parameter to the default value and remove the parameter from subsequent TCD messages, use the **no** form of this command.

cable telco-return spd *number* **dhcp-authenticate**

no cable telco-return spd dhcp-authenticate

Syntax Description

number	Specifies the service provider descriptor (SPD) number.

Defaults

FALSE (0)—Disable the DHCP Authenticate parameter in TCD messages.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Usage Guidelines

The DHCP Authenticate parameter, which is an optional parameter, is expressed as a boolean value.

Examples

The following example sets the DHCP Authenticate parameter:

```
interface cable 6/0
 cable telco-return spd 2 dhcp-authenticate
```

Related Commands

Command	Description
cable telco-return spd dhcp-server	Specifies the IP address of the DHCP server parameter in TCD messages in a cable-routed system.
cable telco-return spd dial-timer	Specifies the Demand Dial Timer TCD parameter for TCD messages in a cable-routed system.
cable telco-return spd factory-default	Indicates the SPD used by cable modems during the factory default procedure.
cable telco-return spd ppp-authenticate	Specifies the PPP Authentication parameter in TCD messages in a cable-routed system.
cable telco-return spd phonenum	Specifies the Telephone Numbers parameter in TCD messages in a cable-routed system.
cable telco-return spd threshold	Specifies the Connection Attempt Threshold parameter for TCD messages in a cable-routed system.

cable telco-return spd dhcp-server

To specify the IP address of the DHCP Server parameter in TCD messages, use the **cable telco-return spd dhcp-server** command in cable interface configuration mode. To set the default value and remove the parameter from subsequent TCD messages, use the **no** form of this command.

cable telco-return spd *number* **dhcp-server** *ip address*

no cable telco-return spd dhcp-server

Syntax Description

spd *number*	Specifies the service provider descriptor (SPD) number.
ip address	Specifies the IP address of the DHCP server.

Defaults

0

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following example specifies the IP address of the DHCP Server parameter in TCD messages:

```
interface cable 6/0
 cable telco-return spd 2 dhcp-server 206.44.207.255
```

Related Commands

Command	Description
cable telco-return spd dhcp-authenticate	Sets the DHCP Authenticate parameter in TCD messages as TRUE (1), which specifies the DHCP server that must be used in a cable-routed system.
cable telco-return spd dial-timer	Specifies the Demand Dial Timer TCD parameter for TCD messages in a cable-routed system.
cable telco-return spd factory-default	Indicates the SPD used by cable modems during the factory default procedure.
cable telco-return spd ppp-authenticate	Specifies the PPP Authentication parameter in TCD messages in a cable-routed system.
cable telco-return spd phonenum	Specifies the Telephone Numbers parameter in TCD messages in a cable-routed system.
cable telco-return spd threshold	Specifies the Connection Attempt Threshold parameter for TCD messages in a cable-routed system.

cable telco-return spd dial-timer

To specify the Demand Dial Timer parameter for TCD messages, use the **cable telco-return spd dial-timer** command in cable interface configuration mode. To set the default value and remove the dial-timer parameter from subsequent TCD messages, use the **no** form of this command.

cable telco-return spd *number* **dial-timer** *seconds*

no **cable telco-return spd dial-timer**

Syntax Description

number	Specifies the service provider descriptor (SPD) number.
seconds	Specifies the Demand Dial Timer parameter for TCD messages in seconds. Valid range is from 0 to 4294967295 seconds.

Defaults

0 seconds

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following example sets the Demand Dial Timer in TCD messages to 7200 seconds (12 minutes):

```
interface cable 6/0
 cable telco-return spd 2 dial-timer 7200
```

Related Commands

Command	Description
cable telco-return spd dhcp-server	Specifies the IP address of the DHCP server parameter in TCD messages in a cable-routed system.
cable telco-return spd dhcp-authenticate	Sets the DHCP Authenticate parameter in TCD messages as TRUE (1), which specifies the DHCP server that must be used in a cable-routed system.
cable telco-return spd factory-default	Indicates the SPD used by cable modems during the factory default procedure.
cable telco-return spd ppp-authenticate	Specifies the PPP Authentication parameter in TCD messages in a cable-routed system.
cable telco-return spd phonenum	Specifies the Telephone Numbers parameter in TCD messages in a cable-routed system.
cable telco-return spd threshold	Specifies the Connection Attempt Threshold parameter for TCD messages in a cable-routed system.

cable telco-return spd factory-default

To indicate the service provider descriptor (SPD) used by cable modems during the factory default procedure, use the **cable telco-return spd factory-default** command in cable interface configuration mode. To restore the default, use the **no** form of this command.

cable telco-return spd *number* **factory-default**

no cable telco-return spd

Syntax Description

number	Specifies the SPD number.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Usage Guidelines

The SPD specified is the factory default SPD.

Examples

The following example indicates the SPD used by cable modem:

```
interface cable 6/0
 cable telco-return spd 2 factory-default
```

Related Commands

Command	Description
cable telco-return spd dhcp-authenticate	Sets the DHCP Authenticate parameter in TCD messages as TRUE (1), which specifies the DHCP server that must be used in a cable-routed system.
cable telco-return spd dhcp-server	Specifies the IP address of the DHCP server parameter in TCD messages in a cable-routed system.
cable telco-return spd dial-timer	Specifies the Demand Dial Timer TCD parameter for TCD messages in a cable-routed system.
cable telco-return spd factory-default	Indicates the SPD used by cable modems during the factory default procedure.
cable telco-return spd ppp-authenticate	Specifies the PPP Authentication parameter in TCD messages in a cable-routed system.

Command	Description
cable telco-return spd phonenum	Specifies the Telephone Numbers parameter in TCD messages in a cable-routed system.
cable telco-return spd threshold	Specifies the Connection Attempt Threshold parameter for TCD messages in a cable-routed system.

cable telco-return spd manual-dial

To specify the Manual Dial parameter in TCD messages, use the **cable telco-return spd manual-dial** command in cable interface configuration mode. To remove the parameter from subsequent TCD messages, use the **no** form of this command.

cable telco-return spd *number* **manual-dial**

no cable telco-return spd *number* **manual-dial**

Syntax Description

number	Specifies the service provider descriptor (SPD) number.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following command specifies that the Manual Dial parameter be included in TCD messages. In this example, the SPD number is 1.

```
interface cable 6/0
 cable telco-return spd 1 manual-dial
```

Related Commands

Command	Description
cable telco-return spd username	Specifies the Login Username parameter in TCD messages in a cable-routed system.
cable telco-return spd password	Specifies the Login Password parameter in TCD messages in a cable-routed system.
cable telco-return spd radius-realm	Specifies the RADIUS Realm SPD parameter in TCD messages in a cable-routed system.

cable telco-return spd password

To specify the login password parameter in TCD messages, use the **cable telco-return spd password** command in cable interface configuration mode. To remove the parameter from subsequent TCD messages, use the **no** form of this command.

cable telco-return spd *number* **password** *string*

no cable telco-return spd password

Syntax Description

number	Specifies the service provider descriptor (SPD) number.
string	Specifies the login password.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following command specifies that the Login Password parameter be included in TCD messages. In this example, the SPD number is 2 and the password is 9JwoKd7.

```
interface cable 6/0
 cable telco-return spd 2 password 9JwoKd7
```

Related Commands

Command	Description
cable telco-return spd service-provider	Specifies the Service Provider Name parameter in TCD messages in a cable-routed system.
cable telco-return spd username	Specifies the Login Username parameter in TCD messages in a cable-routed system.
cable telco-return spd radius-realm	Specifies the RADIUS Realm SPD parameter in TCD messages in a cable-routed system.

cable telco-return spd phonenum

To specify the Telephone Numbers parameter in TCD messages, use the **cable telco-return spd phonenum** command in cable interface configuration mode. To delete any or all previously entered telephone numbers, use the **no** form of this command.

cable telco-return spd *number* **phonenum** *string*

no cable telco-return spd *number* **phonenum**

Syntax Description

number	Specifies the service provider descriptor (SPD) number.
string	Specifies the telephone number to be used by the telephone numbers parameter in TCD messages.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Usage Guidelines

You can repeat this command to enter up to three telephone numbers that are mapped to the telephone numbers parameters in TCD messages. The phone numbers appear as Phone Number 1, Phone Number 2, and Phone Number 3 in the order in which you enter them.

Examples

The following example inserts the telephone number (925) 555-1212 into the TCD messages for SPD 2:

```
interface cable 6/0
 cable telco-return spd 2 phonenum 9255551212
```

Related Commands

Command	Description
cable telco-return spd dhcp-authenticate	Sets the DHCP Authenticate parameter in TCD messages as TRUE (1), which specifies the DHCP server that must be used in a cable-routed system.
cable telco-return spd dhcp-server	Specifies the IP address of the DHCP server parameter in TCD messages in a cable-routed system.
cable telco-return spd dial-timer	Specifies the Demand Dial Timer TCD parameter for TCD messages in a cable-routed system.
cable telco-return spd factory-default	Indicates the SPD used by cable modems during the factory default procedure.
cable telco-return spd ppp-authenticate	Specifies the PPP Authentication parameter in TCD messages in a cable-routed system.
cable telco-return spd threshold	Specifies the Connection Attempt Threshold parameter for TCD messages in a cable-routed system.

cable telco-return spd ppp-authenticate

To specify the PPP Authentication parameter in TCD messages, use the **cable telco-return spd ppp-authenticate** command in cable interface configuration mode. To remove the parameter from subsequent TCD messages, use the **no** form of this command.

cable telco-return spd *number* **ppp-authenticat**e [**both** | **chap** | **pap**]

no cable telco-return spd ppp-authenticate

Syntax Description

number	Specifies the service provider descriptor (SPD) number.
both	(Optional) Specifies both PAP and CHAP authentication.
chap	(Optional) Specifies CHAP authentication.
pap	(Optional) Specifies PAP authentication.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following command specifies that the PPP Authentication parameter be included in TCD messages:

```
interface cable 6/0
 cable telco-return spd 2 ppp-authenticate chap
```

Related Commands

Command	Description
cable telco-return spd dhcp-authenticate	Sets the DHCP Authenticate parameter in TCD messages as TRUE (1), which specifies the DHCP server that must be used in a cable-routed system.
cable telco-return spd dhcp-server	Specifies the IP address of the DHCP server parameter in TCD messages in a cable-routed system.
cable telco-return spd dial-timer	Specifies the Demand Dial Timer TCD parameter for TCD messages in a cable-routed system.
cable telco-return spd factory-default	Indicates the SPD used by cable modems during the factory default procedure.
cable telco-return spd phonenum	Specifies the Telephone Numbers parameter in TCD messages in a cable-routed system.
cable telco-return spd threshold	Specifies the Connection Attempt Threshold parameter for TCD messages in a cable-routed system.

cable telco-return spd radius-realm

To specify the RADIUS Realm parameter in TCD messages, use the **cable telco-return spd radius-realm** command in cable interface configuration mode. To remove the parameter from subsequent TCD messages, use the **no** form of this command.

cable telco-return spd *number* **radius-realm** *string*

no cable telco-return spd *number* **radius-realm**

Syntax Description

number	Specifies the service provider descriptor (SPD) number.
string	Specifies the RADIUS Realm name.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following example activates the RADIUS Realm parameter in TCD messages for SPD 2 and identifies the RADIUS realm named sunol:

```
interface cable 6/0
 cable telco-return spd 2 radius-realm sunol
```

Related Commands

Command	Description
cable telco-return spd service-provider	Specifies the Service Provider Name parameter in TCD messages in a cable-routed system.
cable telco-return spd username	Specifies the Login Username parameter in TCD messages in a cable-routed system.
cable telco-return spd password	Specifies the Login Password parameter in TCD messages in a cable-routed system.

cable telco-return spd service-provider

To specify the Service Provider Name parameter in TCD messages, use the **cable telco-return service-provider** command in cable interface configuration mode. To remove the parameter from subsequent TCD messages, use the **no** form of this command.

cable telco-return spd *number* **service-provider** *string*

no cable telco-return spd *number* **service-provider**

Syntax Description

number	Specifies the service provider descriptor (SPD) number.
string	Specifies the service provider name.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following command specifies that the Service Provider Name parameter be included in TCD messages:

```
interface cable 6/0
 cable telco-return spd 2 service-provider san_jose
```

Related Commands

Command	Description
cable telco-return spd username	Specifies the Login Username parameter in TCD messages in a cable-routed system.
cable telco-return spd password	Specifies the Login Password parameter in TCD messages in a cable-routed system.
cable telco-return spd radius-realm	Specifies the RADIUS Realm SPD parameter in TCD messages in a cable-routed system.

cable telco-return spd threshold

To specify the Connection Attempt Threshold parameter for TCD messages, use the **cable telco-return spd threshold** command for cable interface configuration mode. To set the default threshold number, use the **no** form of this command.

cable telco-return spd *number* **threshold** *number*

no cable telco-return spd threshold

Syntax Description

number	Specifies the service provider descriptor (SPD) number.
number	Specifies the connection attempt threshold. Valid range is from 1 to 255. Default value is 1.

Defaults

1

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following command specifies that the Connection Attempt Threshold parameter be included in TCD messages:

```
interface cable 6/0
 cable telco-return spd 2 threshold 200
```

Related Commands

Command	Description
cable telco-return spd dhcp-authenticate	Sets the DHCP Authenticate parameter in TCD messages as TRUE (1), which specifies the DHCP server that must be used in a cable-routed system.
cable telco-return spd dhcp-server	Specifies the IP address of the DHCP server parameter in TCD messages in a cable-routed system.
cable telco-return spd dial-timer	Specifies the Demand Dial Timer TCD parameter for TCD messages in a cable-routed system.
cable telco-return spd factory-default	Indicates the SPD used by cable modems during the factory default procedure.
cable telco-return spd ppp-authenticate	Specifies the PPP Authentication parameter in TCD messages in a cable-routed system.
cable telco-return spd phonenum	Specifies the Telephone Numbers parameter in TCD messages in a cable-routed system.

cable telco-return spd username

To specify the Login Username parameter in TCD messages, use the **cable telco-return spd username** command in cable interface configuration mode. To remove the parameter from subsequent TCD messages, use the **no** form of this command.

cable telco-return spd *number* **username** *string*

no cable telco-return spd *number* **username**

Syntax Description

number	Specifies the service provider descriptor (SPD) number.
string	Specifies the login username.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following command specifies that the Login Username parameter be included in TCD messages. In this example, the login username is sandy and the SPD is 3.

```
interface cable 6/0
 cable telco-return spd 3 username sandy
```

Related Commands

Command	Description
cable telco-return spd service-provider	Specifies the Service Provider Name parameter in TCD messages in a cable-routed system.
cable telco-return spd password	Specifies the Login Password parameter in TCD messages in a cable-routed system.
cable telco-return spd radius-realm	Specifies the RADIUS Realm SPD parameter in TCD messages in a cable-routed system.

cable time-server

To enable the integrated time-of-day (ToD) server on the Cisco uBR7200 series, enter the **cable time-server** command in global configuration mode. To disable the time-of-day server function, use the **no** form of this command, or enter the command with the **disable** keyword.

cable time-server [**enable** | **disable**]

no cable time-server

Syntax Description

enable	(Optional) Starts the time-of-day server as a background task.
disable	(Optional) Stops the time-of-day server.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following example enables the time-of-day server:

```
cable time-server enable
```

cable upstream admission-control

To specify the percentage of overbooking that will be allowed on the upstream channel, use the **cable upstream admission-control** command in cable interface configuration mode. To disable upstream admission control, use the **no** form of this command.

cable upstream *usport* **admission-control** *percentage*

no cable upstream *usport* **admission-control**

Syntax Description

usport	Specifies the upstream port.
percentage	Sets the admission control as a percentage of the upstream channel capacity. Valid values are from 10 to 1000 percent.

Defaults

Disabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3(6) NA	This command was introduced.

Examples

The following example limits overbooking on upstream port 4 to 500 percent:

```
interface cable 6/0
 cable upstream 4 admission-control 500
```

cable upstream channel-width

To specify an upstream channel width for an upstream port, use the **cable upstream channel-width** command in cable interface configuration mode. To set the channel width back to the default setting of 1600000 Hz, use the **no** form of this command.

cable upstream *usport* **channel-width** *first-choice-width* [*last-choice-width*]

no cable upstream *usport* **channel-width**

Syntax Description

usport	Specifies the port number.
first-choice-width	Specifies upstream channel width in hertz (Hz). Valid values are: 200000 (160000 symbols/sec), 400000 (320000 symbols/sec), 800000 (640000 symbols/sec), 1600000 (1280000 symbols/sec), and 3200000 (2560000 symbols/sec).
last-choice-width	(Optional) The upstream channel width in hertz. The valid values are the same as those for the *first-choice-width* parameter. Use this parameter with the Cisco MC16S cable modem card to enable symbol rate management algorithms. The symbol rate automatically steps up from the *first-choice-width* value to the highest value until a stable channel is established.

Defaults

1600000 Hz

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3(5)NA	This command was introduced.
12.0(4)XI	The *last-choice-width* argument was added.

Usage Guidelines

The *last-choice-width* parameter is only supported by the Cisco MC16S cable modem line card. When the MC16S card is installed, the system attempts to increase the channel width from the *first-choice-width* value to the *last-choice-width* value one step at a time.

Examples

The following example configures upstream port 2 with a channel width of 200,000 Hz (which is equivalent to a symbol rate of 160 kilosymbols/second):

```
interface cable 6/0
 cable upstream 2 channel-width 200000
```

The following example configures upstream port 3 to step from a channel width of 1600000 Hz to a channel width of 3200000 Hz in increments of 200000 Hz:

```
interface cable 6/0
 cable upstream 3 channel-width 1600000 3200000
```

Related Commands

Command	Description
cable upstream fec-strength	Overrides the FEC setting specified in the modulation profile for an upstream channel.
cable upstream hop algorithm	Configures the frequency hop algorithm for the upstream port of a cable router.
cable upstream modulation	Overrides modulation types specified in the modulation profile for the specified upstream channel.

cable upstream data-backoff

To specify automatic or fixed start and stop values for data backoff, use the **cable upstream data-backoff** command in cable interface configuration mode. To use the default data backoff values, use the **no** form of this command.

cable upstream *usport* **data-backoff** {**automatic** | *start end*}

no cable upstream *usport* **data-backoff**

Syntax Description

usport	Specifies the upstream port number.
automatic	Specifies automatic data backoff start and stop values.
start	Binary exponential algorithm. Sets the start value for data backoff. Valid values are from 0 to 15.
end	Binary exponential algorithm. Sets the end value for data backoff. Valid values are from 0 to 15.

Defaults

0 (start), 4 (end)

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

The DOCSIS-specified method of contention resolution for cable modems wishing to send data or requests on the upstream channel is a truncated binary exponential back-off with the initial back-off window and the maximum back-off window controlled by the CMTS. The Cisco uBR7200 series router specifies back-off window values for both data and initial ranging, and sends these values downstream as part of the Bandwidth Allocation Map (MAP) MAC message. The values are power-of-two values. For example, a value of 4 indicates a window between 0 and 15; a value of 10 indicates a window between 0 and 1023.

Cisco recommends that you use the automatic settings for data backoff.

Examples

The following example sets the automatic data backoff values for port 2:

```
interface cable 6/0
 cable upstream 2 data-backoff automatic
```

cable upstream fec

To enable upstream forward error correction (FEC), use the **cable upstream fec** command in cable interface configuration mode. To disable FEC, use the **no** form of this command.

cable upstream *usport* **fec**

no cable upstream *usport* **fec**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.

Defaults

Enabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

The Cisco uBR7200 series uses forward error correction (FEC) to attempt to correct any upstream data that might have been corrupted. To use this feature, you need to activate FEC on the upstream RF carrier. When FEC is activated, the Cisco uBR7200 series commands all cable modems on the network to activate FEC.

Examples

The following example activates upstream forward error correction:

```
interface cable 6/0
 cable upstream 0 fec
```

Related Commands

Command	Description
cable upstream frequency	Enters a fixed frequency of the upstream RF carrier for an upstream port.
cable upstream power-level	Sets the input power level for the upstream RF carrier in decibels per millivolt (dBmV).
cable upstream scrambler	Enables the cable upstream scrambler.
cable upstream shutdown	Disables the upstream port.

cable upstream fec-strength

To override the forward error correction (FEC) setting specified in the modulation profile for this upstream channel, use the **cable upstream fec-strength** command in cable interface configuration mode. To restore the default value, use the **no** form of this command.

cable upstream *usport* **fec-strength** *t-bytes*

no cable upstream *usport* **fec-strength**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.
t-bytes	Overrides the FEC strength specified in the modulation profile for this upstream channel. Valid values are from 0 to 10, where: • 0 disables FEC. • 1 is the lowest FEC strength. • 10 is the highest FEC strength.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following example shows how to comfigure the **cable upstream fec-strength** command:

```
interface cable 6/0
 cable upstream 2 fec-strength 3
```

Related Commands

Command	Description
cable upstream fec	Enables the upstream FEC.
cable upstream channel-width	Specifies an upstream channel width for a headend cable router.
cable upstream hop algorithm	Configures the frequency hop algorithm for the upstream port of a cable router.
cable upstream modulation	Overrides modulation types specified in the modulation profile for the specified upstream channel.

cable upstream frequency

To enter a fixed frequency of the upstream radio frequency (RF) carrier for an upstream port, use the **cable upstream frequency** command in cable interface configuration mode. To restore the default value for this command, use the **no** form of this command.

cable upstream *usport* **frequency** *up-freq-hz*

no cable upstream *usport* **frequency** *up-freq-hz*

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.
up-freq-hz	The upstream center frequency is configured to a fixed value. The valid range is from 5,000,000 to 42,000,000 Hz.

Defaults

Upstream center frequency is not configured to a fixed value.

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

The upstream channel frequency of your RF output must be set to comply with the expected input frequency of your Cisco cable modem line card. To configure an upstream channel frequency, you may:

- Configure a fixed frequency between 5 to 42 MHz and enable the upstream port, or
- Create a global spectrum group, assign the interface to it, and enable the upstream port.

To configure the default upstream frequency (which is no fixed frequency), enter the **cable upstream** *usport* **frequency** command without specifying a center frequency.

Examples

The following example configures the upstream center frequency for port 0 (located in slot 6) to 5700000 Hz:

```
interface cable 6/0
 cable upstream 0 frequency 5700000
```

cable upstream freq-adj averaging

To control power adjustments on a Cisco uBR7200 series cable router by setting the frequency threshold, use the **cable upstream freq-adj averaging** interface configuration command. To disable power adjustments, use the **no** form of this command.

cable upstream *n* **freq-adj averaging** *% of frequency adjustment*

no cable upstream freq-adj averaging

Syntax Description

n	Specifies the upstream port number.
averaging	Specifies that a percentage of frequency adjustment packets is required to change the adjustment method from the regular power adjustment method to the noise power adjustment method.
% of frequency adjustment	Specifies the percentage of frequency adjustment packets required to switch from the regular power adjustment method to the noise power adjustment method. Valid range is from 10 to 100%.

Defaults

No default behavior or values.

Command Modes

Interface configuration

Command History

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

Examples

The following example illustrates how to change the power adjustment method when the frequency adjustment packet count reaches 50 percent:

```
cable upstream 0 freq-adj averaging 50
```

Related Commands

Command	Description
cable upstream power-adjust	Controls power adjustment methods on the Cisco uBR7200 series cable routers.
show cable flap-list	Displays a list of cable modems that have exceeded the threshold number of power adjustments.
show cable modem	Displays cable modem configuration settings.

cable upstream hop algorithm

To configure the frequency hop algorithm for the upstream port, use the **cable upstream hop algorithm** command in cable interface configuration mode. To configure the optimum algorithm, use the **no** form of this command.

cable upstream *usport* **hop algorithm** {**blind** | **optimum**}

no cable upstream *usport* **hop algorithm**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.
blind	Selects the blind frequency hop algorithm.
optimum	Selects the optimum hop algorithm.

Defaults

Optimum

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Usage Guidelines

This command is only supported on the Cisco MC16S cable modem card.

Examples

The following example configures the optimum hop algorithm:

```
interface cable 6/0
 cable upstream 3 hop algorithm optimum
```

Related Commands

Command	Description
cable upstream channel-width	Specifies an upstream channel width for a headend cable router.
cable upstream fec-strength	Overrides the FEC setting specified in the modulation profile for an upstream channel.
cable upstream modulation	Overrides modulation types specified in the modulation profile for the specified upstream channel.

cable upstream minislot-size

To specify the minislot size (in ticks) for a specific upstream interface, use the **cable upstream minislot-size** command in cable interface configuration mode. To set the default minislot size of 8 if this is valid for the current channel width setting, use the **no** form of this command.

cable upstream *usport* **minislot-size** *size*

no cable upstream *usport* **minislot-size**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.
size	Specifies the minislot size in time ticks. Valid minislot sizes are: 2 (32 symbols), 4 (64 symbols), 8 (128 symbols), 16 (256 symbols), 32 (512 symbols), 64 (1024 symbols), and 128 (2048 symbols).

Defaults

8

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3(6) NA	This command was introduced.

Usage Guidelines

Caution Using values of 64 or 128 for higher symbol rates such as 1280 kilosymbols/second or 2560 kilosymbols/second can cause performance problems. Depending on your current setting's symbol rate, you should select the minislot size (in ticks) that yields a minislot size of 32 or 64 symbols.

Examples

The following example sets the minislot size on upstream port 4 to 16 (or 256 symbols):

```
interface cable 6/0
 cable upstream 4 minislot-size 16
```

cable upstream modulation

To override modulation types specified in the modulation profile for the specified upstream channel, use the **cable upstream modulation** command in cable interface configuration mode. To disable overriding modulation profiles, use the **no** form of this command.

cable upstream *usport* **modulation** *first-choice-mod* [*last-choice-mod*]

no cable upstream *usport* **modulation**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.
first-choice-mod	Overrides the modulation type specified in the modulation profile for this channel. Valid values are qpsk or qam16.
last-choice-mod	(Optional) Valid values are qpsk or qam16. Make sure this parameter is different from the *first-choice-mod* setting or it will be ignored. When used, this parameter enables modulation management algorithms.

Defaults

No default behavior or values.

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Usage Guidelines

This feature is only supported on the Cisco MC16S cable modem card.

The Cisco uBR7200 series cable router first attempts to apply the modulation type specified in the first choice parameter. Then the cable router attempts to apply the modulation type specified in the second choice parameter. Whenever possible, the cable router tries to raise the modulation type to qam16.

Examples

The following example shows how to configure the **cable upstream modulation** command:

```
interface cable 6/0
 cable upstream 3 modulation qpsk qam16
```

Related Commands

Command	Description
cable upstream modulation-profile	Assigns a modulation profile to an interface on a cable router.
cable upstream channel-width	Specifies an upstream channel width for a headend cable router.
cable upstream fec-strength	Overrides the FEC setting specified in the modulation profile for an upstream channel.
cable upstream hop algorithm	Configures the frequency hop algorithm for the upstream port of a cable router.

cable upstream modulation-profile

To assign a modulation profile to an upstream interface, use the **cable upstream modulation-profile** command in cable interface configuration mode. To assign modulation profile 1 to the interface, use the **no** form of this command.

cable upstream *usport* **modulation-profile** *profile*

no cable upstream *usport* **modulation-profile**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.
profile	Assigns the modulation profile to the specified interface.

Defaults

Modulation profile 1

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following example assigns modulation profile 8 to upstream port 2:

```
interface cable 6/0
 cable upstream 2 modulation-profile 8
```

Related Commands

Command	Description
cable modulation-profile	Defines the modulation profile for a cable router.

cable upstream power-adjust

To control power adjustment methods on the Cisco uBR7200 series cable routers, use the **cable upstream power-adjust** command in interface configuration mode. To disable power adjustments, use the **no** form of this command.

cable upstream *n* **power-adjust** [**continue**] [**noise** *% of power adjustment*] [**threshold** *#*]

no cable upstream power-adjust

Syntax Description

n	Specifies the upstream port number.
continue	Specifies the regular power adjustment method (minimum power adjustments).
noise	Specifies that a percentage of power adjustment packets is required to change the adjustment method from the regular power adjustment method to the noise power adjustment method.
% of power adjustment	Specifies the percentage of power adjustment packets required to switch from the regular power adjustment method to the noise power adjustment method. Valid range is from 10 to 100%.
threshold *#*	Specifies the power adjustment threshold. The threshold range is from 0 to 10dB.

Defaults

No default behavior or values.

Command Modes

Interface configuration

Command History

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

Examples

The following example illustrates how to change the power adjustment method when the percentage of power adjustment packets reaches 50 percent:

```
cable upstream 0 power-adjust noise 50
```

Related Commands

Command	Description
cable upstream frequency	Configures a fixed frequency of the upstream RF carrier for an upstream port.

Command	Description
show cable flap-list	Displays a list of cable modems that have exceeded the threshold number of power adjustments.
show cable modem	Displays cable modem configuration settings.

cable upstream power-level

To set the input power level for the upstream radio frequency (RF) carrier in decibels per millivolt (dBmV), use the **cable upstream power-level** command in cable interface configuration mode. To restore the default value for this command, use the **no** form of this command.

cable upstream *usport* **power-level** *dbmv*

no cable upstream *usport* **power-level** *dbmv*

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.
dbmv	Decibels per millivolt designating the upstream signal input power level. Valid range is –10 dBmV to 25 dBmV.

Defaults

0 dBmV

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

The Cisco uBR7200 series controls the output power levels of the cable modems to meet the desired upstream input power level. The nominal input power level for the upstream RF carrier is specified in decibels per millivolt (dBmV). The default setting of 0 dBmV is the optimal setting for the upstream power level.

The valid range for the input power level depends on the data rate. At 1.6 MHz, the valid range is –10 dBmV to 25 dBmV. Higher values cause the modems to increase their transmit power, achieving a greater carrier-to-noise ratio (CNR). If your power levels operate at greater than the maximum valid level, you must use an inline attenuator to bring the power level to within the valid range.

Caution If you increase the input power level, the cable modems on your HFC network will increase their transmit power level. This might cause an increase in the carrier-to-noise ratio (CNR) on the network. Be careful if you adjust this parameter. You might violate the upstream return laser design parameters.

You should not adjust your input power level by more than 5 dB in a 30-second interval. If you *increase* the power level by more than 5 dB within 30 seconds, cable modem service on your network will be disrupted. If you *decrease* the power level by more than 5 dB within 30 seconds, cable modems on your network will be forced offline.

When you run **cable upstream 0 power-level**, Cisco recommends that the adjacent channel not have a large variation. The recommended maximum input power variance is 5 to 6 dBmV.

Examples

The following example sets the input power level for upstream port 0 to -5 dBmV:

```
interface cable 6/0
 cable upstream 0 power-level -5
```

Related Commands

Command	Description
cable upstream fec	Enables the upstream FEC.
cable upstream frequency	Enters a fixed frequency of the upstream RF carrier for an upstream port.
cable upstream scrambler	Enables the cable upstream scrambler.
cable upstream shutdown	Disables the upstream port.

cable upstream range-backoff

To specify automatic or configured initial ranging backoff calculation, use the **cable upstream range-backoff** command in cable interface configuration mode. To set default values, use the **no** form of this command.

cable upstream *usport* **range-backoff** {**automatic** | *start end*}

no cable upstream *usport* **range-backoff**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.
automatic	Specifies the fixed data backoff start and end values.
start	Binary exponential algorithm. Sets the start value for initial ranging backoff. Valid values are from 0 to 15.
end	Binary exponential algorithm. Sets the end value for initial ranging backoff. Valid values are from 0 to 15.

Defaults

0 (start), 4 (end)

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

The DOCSIS-specified method of contention resolution for cable modems wishing to send data or requests on the upstream channel is a truncated binary exponential back-off with the initial back-off window and the maximum back-off window controlled by the CMTS. The Cisco uBR7200 series router specifies back-off window values for both data and initial ranging, and sends these values downstream as part of the Bandwidth Allocation Map (MAP) MAC message. The values are power-of-two values. For example, a value of 4 indicates a window between 0 and 15; a value of 10 indicates a window between 0 and 1023.

The **automatic** setting is optimized for up to 250 cable modems per upstream port. Set manual values for data back-off windows only when operating with more than 250 cable modems per upstream port.

Examples

The following example sets the range backoff to **automatic** for upstream port 2:

```
interface cable 6/0
 cable upstream 2 range-backoff automatic
```

cable upstream rate-limit

To set DOCSIS rate limiting for an upstream port on a cable modem card, use the **cable upstream rate-limit** command in cable interface configuration mode. To disable DOCSIS rate limiting for the upstream port, use the **no** form of this command.

cable upstream *usport* **rate-limit** [**token-bucket** [**shaping**]]

no cable upstream *usport* **rate-limit**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.
token-bucket	(Optional) Enables rate limiting for the specified upstream cable interface using the token bucket policing algorithm.
shaping	(Optional) Enables rate limiting for the specified upstream cable interface using the token bucket policing algorithm with traffic shaping.

Defaults

Token bucket algorithm with traffic shaping.

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3(6)NA	This command was introduced.
11.3(9)NA	The **shaping** keyword was added.

Usage Guidelines

Upstream rate limiting allows upstream bandwidth requests from rate-exceeding cable modems to be buffered without incurring TCP-related timeouts and retransmits. This enables the Cisco uBR7200 series to enforce the peak upstream rate for each cable modem without degrading overall TCP performance for the subscriber CPEs. Upstream grant shaping is per cable modem (SID).

When the **token-bucket** algorithm is configured, the Cisco uBR7200 series will automatically drop packets in violation of allowable upstream bandwidth.

Use of the default value (the upstream port's rate limit) enforces strict DOCSIS-compliant rate limiting.

Examples

The following example configures the token bucket filter algorithm with traffic shaping on upstream port 4:

```
interface cable 6/0
 cable upstream 4 rate-limit token-bucket
```

Related Commands

Command	Description
cable downstream rate-limit	Enables DOCSIS rate limiting on downstream traffic.

cable upstream scrambler

To enable the cable upstream scrambler, use the **cable upstream scrambler** command in cable interface configuration mode. To restore the default configuration value for this command, use the **no** form of this command.

cable upstream *usport* **scrambler**

no cable upstream *usport* **scrambler**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.

Defaults

Disabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

This command causes cable modems to enable their pseudo-random scrambler circuitry to improve the robustness of the upstream receiver on the line card.

The scrambler on the upstream RF carrier enables cable modems on the HFC network to use built-in scrambler circuitry for upstream data transmissions. The scrambler circuitry improves reliability of the upstream receiver on the cable modem card. The upstream scrambler is activated by default and should not be disabled under normal circumstances.

Caution Scrambler must be activated for normal operation. Deactivate only for prototype modems that do not support scrambler.

Examples

The following example activates the upstream scrambler:

```
interface cable 6/0
 cable upstream 0 scrambler#
```

Related Commands

Command	Description
cable upstream fec	Enables the upstream FEC.
cable upstream frequency	Enters a fixed frequency of the upstream RF carrier for an upstream port.
cable upstream power-level	Sets the input power level for the upstream RF carrier in decibels per millivolt (dBmV).
cable upstream shutdown	Disables the upstream port.

cable upstream shutdown

To disable the upstream port, use the **cable upstream shutdown** command in cable interface configuration mode. To enable the upstream port, use the **no** form of this command.

cable upstream *usport* **shutdown**

no cable upstream *usport* **shutdown**

Syntax Description

usport	Specifies the upstream port number. Valid range is from 0 to 5 if you are using a Cisco MC16 cable modem card.

Defaults

Upstream port enabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 XA	This command was introduced.

Examples

The following example disables the upstream port:

```
interface cable 6/0
 cable upstream 0 shutdown
```

cable-modem compliant bridge

To enable DOCSIS-compliant transparent bridging for a cable access router interface at startup, use the **cable-modem compliant bridge** command in cable interface configuration command. To disable DOCSIS-compliant bridging for the interface, use the **no** form of this command.

cable-modem compliant bridge

no cable-modem compliant bridge

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

It is normally not necessary to enter this command in data-over-cable bridging applications because DOCSIS-compliant bridging is enabled by default. If you wish to do full transparent bridging rather than DOCSIS-compliant bridging, use the **no** form of the command, then configure full transparent bridging using CLI commands.

Examples

The following example shows how to enter the **cable-modem compliant bridge** command for a cable access router interface, starting from global configuration mode:

```
interface cable-modem 0
 cable-modem compliant bridge
```

Related Commands

Command	Description
cable-modem downstream saved channel	Modifies the saved downstream channel setting and upstream power value on a cable access router interface.
cable-modem fast-search	Enables a faster downstream search algorithm on a cable access router interface.
cable-modem upstream preamble qpsk	Enables the QPSK modulation scheme in the upstream direction from the cable modem interface to the headend.
cable-modem voip best-effort	Allows voice calls to be sent upstream over the cable interface via best effort.

cable-modem downstream saved channel

To modify the saved downstream channel setting and upstream power value on a cable access router interface, use the **cable-modem downstream saved channel** command in cable interface configuration mode. To remove the saved settings, which will be resaved at the next initialization cycle, use the **no** form of this command.

cable-modem downstream saved channel *ds-frequency us-power*

no cable-modem downstream saved channel *ds-frequency us-power*

Syntax Description

ds-frequency	Downstream channel frequency in Hz, which can be from 91000000 to 860000000.
us-power	Upstream power level in decibels per millivolt (dBmV), which can be from 8 to 61.

Defaults

Enabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

This command is auto-generated by the operation of the cable MAC layer process. The DOCSIS RFI specification requires that cable modems remember the downstream frequency and upstream power of the last successfully ranged session. These parameters are called up as the first downstream frequency and upstream power to use the next time the cable modem is booted. This operation dramatically speeds up the channel search.

Use the **no cable-modem downstream saved channel** *ds-frequency us-power* command to remove the saved frequency and power setting from the running configuration, which will be resaved at the next initialization cycle.

Cisco recommends that this command NOT be used by end users of the Cisco uBR924 cable access router.

Examples

The following example shows how to remove the downstream frequency of 91000000 Hz and the upstream power level of 33 dBmV from the running configuration of a cable-modem interface, starting from global configuration mode:

```
interface cable-modem 0
 no cable-modem downstream saved channel 91000000 33
```

Related Commands

Command	Description
cable max-hosts	Enables DOCSIS-compliant transparent bridging for a cable modem interface at startup.
cable-modem fast-search	Enables a faster downstream search algorithm on a cable access router interface.
cable-modem upstream preamble qpsk	Enables the QPSK modulation scheme in the upstream direction from the cable modem interface to the headend.
cable-modem voip best-effort	Allows voice calls to be sent upstream over the cable interface via best effort.

cable-modem fast-search

To enable a faster downstream search algorithm on a cable access router interface, use the **cable-modem fast-search** command in cable interface configuration mode. To disable the downstream fast-search feature, use the **no** form of this command.

cable-modem fast-search

no cable-modem fast-search

Syntax Description

There are no keywords or arguments for this command.

Defaults

Disabled

Command Modes

Cable interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

This feature speeds up the frequency search performed by the cable access router. Normally it takes the cable access router about 30 to 50 seconds to sample 30 to 50 frequencies. The **cable-modem fast-search** command can reduce this search time. However, there might be some cases where this fast-search algorithm might not perform as well as the default algorithm. Trial and error is the only way to discover how well this feature works for your environment.

Examples

The following example shows how to enter the **cable-modem fast-search** command, beginning in global configuration mode:

```
interface cable-modem 0
 cable-modem fast-search
```

Related Commands

Command	Description
cable max-hosts	Enables DOCSIS-compliant transparent bridging for a cable modem interface at startup.
cable-modem downstream saved channel	Modifies the saved downstream channel setting and upstream power value on a cable access router interface.
cable-modem upstream preamble qpsk	Enables the QPSK modulation scheme in the upstream direction from the cable modem interface to the headend.
cable-modem voip best-effort	Allows voice calls to be sent upstream over the cable interface via best effort.

cable-modem upstream preamble qpsk

To enable the QPSK modulation scheme in the upstream direction from the cable access router interface to the headend, use the **cable-modem upstream preamble qpsk** command in cable interface configuration mode. To disable upstream modulation for the interface, use the **no** form of this command.

cable-modem upstream preamble qpsk

no cable-modem upstream preamble qpsk

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Interface configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following example shows how to enter the **cable-modem upstream preamble qpsk** command for a cable access router interface, beginning in global configuration mode:

```
interface cable-modem 0
 cable-modem upstream preamble qpsk
```

Related Commands

Command	Description
cable max-hosts	Enables DOCSIS-compliant transparent bridging for a cable modem interface at startup.
cable-modem downstream saved channel	Modifies the saved downstream channel setting and upstream power value on a cable access router interface.
cable-modem fast-search	Enables a faster downstream search algorithm on a cable access router interface.
cable-modem voip best-effort	Allows voice calls to be sent upstream over the cable interface via best effort.

cable-modem voip best-effort

To allow voice calls to be sent upstream over the cable interface via best effort, use the **cable-modem voip best-effort** command in cable interface configuration mode. To disable best-effort voice calls, use the **no** form of this command.

cable-modem voip best-effort

no cable-modem voip best-effort

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Cable interface configuration

Command History

Release	Modification
12.0(5)T	This command was introduced.

Usage Guidelines

This command allows you to configure the voice traffic on a Cisco uBR924 to allow only calls having a high priority service identifier (SID) to be connected.

If the dynamic configuration of high priority queues for voice traffic fails, or if the far end cannot support the multiple SIDs and multiple classes of service required by high priority traffic, the flag set by this command will be checked. If enabled (the default setting), the call will be allowed to go through. If disabled, the call will fail.

Examples

The following example shows how to disable best-effort voice calls on a Cisco uBR924 cable interface beginning in global configuration mode:

```
interface cable-modem 0
 no cable-modem voip best-effort
```

Related Commands

Command	Description
cable max-hosts	Enables DOCSIS-compliant transparent bridging for a cable modem interface at startup.
cable-modem downstream saved channel	Modifies the saved downstream channel setting and upstream power value on a cable access router interface.
cable-modem fast-search	Enables a faster downstream search algorithm on a cable access router interface.
cable-modem upstream preamble qpsk	Enables the QPSK modulation scheme in the upstream direction from the cable modem interface to the headend.

call application voice

To create an application and to indicate the location where the corresponding TCL files that implement this application are located, use the **call application voice** command in global configuration mode. To remove the defined application and all configured parameters associated with it, use the **no** form of the command.

call application voice *application-name location*

no call application voice *application-name location*

Syntax Description

application-name	Character string that defines the name of the application.
location	The location of the TCL file in URL format. Valid storage locations are TFTP, FTP, and Flash.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use this command when configuring interactive voice response (IVR) or one of the IVR-related features (such as Debit Card) to define the name of an application and to identify the location of the TCL script associated with this application.

Examples

This example shows how to define the application "prepaid" and the TFTP server location of the associated TCL script:

```
call application voice prepaid tftp://keyer/debitcard.tcl
```

Related Commands

Command	Description
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reload the designated TCL script.
call application voice pin-len	Defines the number of characters in the personal identification number (PIN) for the application and passes that information to the application.

Command	Description
call application voice redirect-number	Defines the telephone number to which a call will be redirected—for example, the operator telephone number of the service provider—for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines the number of seconds a user is warned before their allowed calling time runs out for the designated application.

call application voice language

To define the language of the audio file for the specified application and to pass that information to the specified application, use the **call application voice language** command in global configuration mode. To remove the associated language of the audio file from the application, use the **no** form of this command.

call application voice *application-name* **language** *number language*

no call application voice *application-name* **language** *number language*

Syntax Description

application-name	The name of the application to which the language parameters are being passed.
number	Tag that uniquely identifies an audio file. Valid entries are 0 to 9.
language	Defines the language of the associated audio file. Valid entries are: • **en**—English • **sp**—Spanish • **ch**—Mandarin • **aa**—all

Defaults

No default behavior or values.

Command Modes

Global configuration mode

Command History

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

Usage Guidelines

Use this command when configuring IVR (depending on the TCL script being used) or one of the IVR-related features (such as Debit Card) to define the language of the audio file for the specified application and to pass that information to the specified application.

Table 4 lists TCL script names and the corresponding parameters that are required for each TCL scripts.

Table 4 TCL Scripts and Parameters

TCL Script Name	Description —Summary	Parameters to Configure
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. The length of digits allowed for the account number and password are configurable through the command-line interface (CLI). If the authentication fails, it allows the caller to retry. The retry number is also configured through the CLI.	• **call application voice uid-len** min = 1, max = 20, default = 10 • **call application voice pin-len** min = 0, max = 10, default = 4 • **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account and PIN respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and pin respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	• **call application voice retry-count** min = 1, max = 5, default = 3

Examples

The following example shows how to define English and Spanish as the languages of the audio files associated with the application named prepaid:

```
call application voice prepaid language 1 en
call application voice prepaid language 2 sp
```

Related Commands

Command	Description
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice load	Reload the designated TCL script.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Defines the telephone number to which a call will be redirected—for example, the operator telephone number of the service provider—for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines the number of seconds a user is warned before their allowed calling time runs out for the designated application.

call application voice load

To reload the selected TCL script from the URL, use the **call application voice load** command in privileged EXEC mode.

call application voice load *name*

Syntax Description

name	Defines the TCL application to use for the call.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The software checks the signature lock to ensure it is a Cisco-supported TCL script.

Note If the TCL script does not have a valid Cisco-supported signature, the software fails to load the script and generates the following error message:

```
00:02:54: %IVR-3-BAD_IVR_SIG: Script signature is invalid
```

Examples

The following example shows how to reload the TCL script called clid_4digits_npw_3.tcl:

```
call application voice load clid_4digits_npw_3.tcl
```

call application voice pin-len

To define the number of characters in the personal identification number (PIN) for the designated application, use the **call application voice pin-len** command in global configuration mode. To restore default values for this command, use the **no** form of this command.

call application voice *application-name* **pin-len** *number*

no call application voice *application-name* **pin-len** *number*

Syntax Description

application-name	The name of the application to which the PIN length parameter is being passed.
number	Defines the number of allowable characters in PINs associated with the specified application. Valid entries are 0 to 10.

Defaults

No default behavior or values.

Command Modes

Global configuration mode

Command History

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

Usage Guidelines

Use this command when configuring IVR (depending on the TCL script being used) or one of the IVR-related features (such as Debit Card) to define the number of allowable characters in a PIN for the specified application and to pass that information to the specified application.

Table 5 lists TCL script names and the corresponding parameters that are required for each TCL scripts.

Table 5 TCL Scripts and Parameters

TCL Script Name	Description—Summary	Parameters to Configure
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. The length of digits allowed for the account number and password are configurable through the command-line interface (CLI). If the authentication fails, it allows the caller to retry. The retry number is also configured through the CLI.	• **call application voice uid-len** min = 1, max = 20, default = 10 • **call application voice pin-len** min = 0, max = 10, default = 4 • **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account and PIN respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and pin respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	• **call application voice retry-count** min = 1, max = 5, default = 3

Examples

The following example shows how to define a PIN length of 4 characters for the application named prepaid:

```
call application voice prepaid pin-len 4
```

Related Commands

Command	Description
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reload this designated TCL script.
call application voice redirect-number	Defines the telephone number to which a call will be redirected—for example, the operator telephone number of the service provider—for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines the number of seconds a user is warned before their allowed calling time runs out for the designated application.

call application voice redirect-number

To define the telephone number to which a call will be redirected—for example, the operator telephone number of the service provider—for the designated application, use the **call application voice redirect-number** command in global configuration mode. To cancel this particular parameter, use the **no** form of this command.

call application voice *application-name* **redirect-number** *number*

no call application voice *application-name* **redirect-number** *number*

Syntax Description

application-name	The name of the application to which the redirect telephone number parameter is being passed.
number	Defines the designated operator telephone number of the service provider (or any other number designated by the customer). This is the number that calls are terminated to when, for example, debit time allowed has run out or the debit amount is exceeded.

Defaults

No default behavior or values.

Command Modes

Global configuration mode

Command History

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

Usage Guidelines

Use this command when configuring IVR (depending on the TCL script being used) or one of the IVR-related features (such as Debit Card) to define the telephone number to which a call will be redirected.

Table 6 lists TCL script names and the corresponding parameters that are required for each TCL scripts.

Table 6 TCL Scripts and Parameters

TCL Script Name	Description —Summary	Parameters to Configure
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. The length of digits allowed for the account number and password are configurable through the command-line interface (CLI). If the authentication fails, it allows the caller to retry. The retry number is also configured through the CLI.	• **call application voice uid-len** min = 1, max = 20, default =10 • **call application voice pin-len** min = 0, max = 10, default = 4 • **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account and PIN respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and pin respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	• **call application voice retry-count** min = 1, max = 5, default = 3

Examples

The following example shows how to define a redirect number for the application named prepaid:

```
call application voice prepaid redirect-number 5551111
```

Related Commands

Command	Description
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reload the designated TCL script.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines the number of seconds a user is warned before their allowed calling time runs out for the designated application.

call application voice retry-count

To define the number of times a caller is permitted to reenter the personal identification number (PIN) for the designated application, use the **call application voice retry-count** command in global configuration mode. To cancel this particular parameter, use the **no** form of this command.

call application voice *application-name* **retry-count** *number*

no call application voice *application-name* **retry-count** *number*

Syntax Description

application-name	The name of the application to which the number of possible retries is being passed.
number	Defines the number of times the caller is permitted to re-enter PIN digits. Valid entries for this parameter are 1 to 5.

Defaults

No default behavior or values.

Command Modes

Global configuration mode

Command History

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

Usage Guidelines

Use this command when configuring IVR (depending on the TCL script being used) or one of the IVR-related features (such as Debit Card) to define how many times a user can reenter a PIN.

Table 7 lists TCL script names and the corresponding parameters that are required for each TCL scripts.

Table 7 TCL Scripts and Parameters

TCL Script Name	Description —Summary	Parameters to Configure
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. The length of digits allowed for the account number and password are configurable through the command-line interface (CLI). If the authentication fails, it allows the caller to retry. The retry number is also configured through the CLI.	• **call application voice uid-len** min = 1, max = 20, default =10 • **call application voice pin-len** min = 0, max = 10, default = 4 • **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account and PIN respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and pin respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	• **call application voice retry-count** min = 1, max = 5, default = 3

Examples

The following example shows how to define that a user can re-enter a PIN 3 times before being disconnected for the application named prepaid:

```
call application voice prepaid retry-count 3
```

Related Commands

Command	Description
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reload the designated TCL script.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Defines the telephone number to which a call will be redirected—for example, the operator telephone number of the service provider—for the designated application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines the number of seconds a user is warned before their allowed calling time runs out for the designated application.

call application voice set-location

To define the location, language, and category of the audio files for the specified application, use the **call application voice set-location** command in global configuration mode. To cancel this particular parameter, use the **no** form of this command.

call application voice *application-name* **set-location** *language category location*

no call application voice *application-name* **set-location** *language category location*

Syntax Description

application-name	The name of the application to which the set-location parameters are being passed.
language	Defines the language associated with the audio files. Possible values for this parameter are: • **en** = English, • **ch** = Mandarin • **sp** = Spanish
category	Defines a particular category group. Audio files can be divided into category groups (from 0 to 4). For example, audio files representing the days and months can be category 1, audio files representing units of currency can be category 2, audio files representing units of time: seconds, minutes, and hours can be category 3. Min = 0, Max = 4 (0 means all).
location	Defines the location (audio file URL or directory in the TFTP server) where the audio files are stored.

Defaults

No default behavior or values.

Command Modes

Global configuration mode

Command History

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

Usage Guidelines

Use this command when configuring IVR (depending on the TCL script being used) or one of the IVR-related features (such as Debit Card) to define the location, language, and category of the audio files for the designated application and pass that information to the application.

Table 8 lists TCL script names and the corresponding parameters that are required for each TCL scripts.

Table 8 TCL Scripts and Parameters

TCL Script Name	Description —Summary	Parameters to Configure
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. The length of digits allowed for the account number and password are configurable through the command-line interface (CLI). If the authentication fails, it allows the caller to retry. The retry number is also configured through the CLI.	• **call application voice uid-len** min = 1, max = 20, default =10 • **call application voice pin-len** min = 0, max = 10, default = 4 • **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account and PIN respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and pin respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	• **call application voice retry-count** min = 1, max = 5, default = 3

Examples

The following example shows how to configure the **call application voice set-location** command for the application named prepaid. In this example, the language defined is English, the category into which the audio files are group is Category 0 (meaning all) and the location is the keyer directory on the TFTP server.

```
call application voice prepaid set-location en 0 tftp://keyer/
```

Related Commands

Command	Description
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reload this designated TCL script.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Defines the telephone number to which a call will be redirected—for example, the operator telephone number of the service provider—for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application.
call application voice warning-time	Defines the number of seconds a user is warned before their allowed calling time runs out for the designated application.

call application voice uid-len

To define the number of characters in the user identification number (UID) for the designated application, use the **call application voice uid-length** command in global configuration mode. To restore default values for this command, use the **no** form of this command.

call application voice *application-name* **uid-len** *number*

no call application voice *application-name* **uid-len** *number*

Syntax Description

application-name	The name of the application to which the UID length parameter is being passed.
number	Defines the number of allowable characters in UIDs associated with the specified application. Valid entries are 1 to 20.

Defaults

No default behavior or values.

Command Modes

Global configuration mode

Command History

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

Usage Guidelines

Use this command when configuring IVR (depending on the TCL script being used) or one of the IVR-related features (such as Debit Card) to define the number of allowable characters in a UID for the specified application and to pass that information to the specified application.

Table 9 lists TCL script names and the corresponding parameters that are required for each TCL scripts.

Table 9 TCL Scripts and Parameters

TCL Script Name	Description —Summary	Parameters to Configure
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. The length of digits allowed for the account number and password are configurable through the command-line interface (CLI). If the authentication fails, it allows the caller to retry. The retry number is also configured through the CLI.	• **call application voice uid-len** min = 1, max = 20, default =10 • **call application voice pin-len** min = 0, max = 10, default = 4 • **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account and PIN respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and pin respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	• **call application voice retry-count** min = 1, max = 5, default = 3

Examples

The following example shows how to configure 4 allowable characters in the UID for the application named prepaid:

```
call application voice prepaid uid-len 4
```

Related Commands

Command	Description
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reload this designated TCL script.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Defines the telephone number to which a call will be redirected—for example, the operator telephone number of the service provider—for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice warning-time	Defines the number of seconds a user is warned before their allowed calling time runs out for the designated application.

call application voice warning-time

To define the number of seconds a user is warned before the allowed calling time runs out, use the **call application voice warning-time** command in global configuration mode. To restore default values for this command, use the **no** form of this command.

call application voice *application-name* **warning-time** *number*

no call application voice *application-name* **warning-time** *number*

Syntax Description

application-name	The name of the application to which the warning time parameter is being passed.
number	Defines the number of seconds the user is warned before the allowed calling time runs out. Valid entries are 10 to 600.

Defaults

No default behavior or values.

Command Modes

Global configuration mode

Command History

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

Usage Guidelines

Use this command when configuring IVR (depending on the TCL script being used) or one of the IVR-related features (such as Debit Card) to define the number of seconds a user is warned before the allowed calling time runs out for the specified application and to pass that information to the specified application.

Table 10 lists TCL script names and the corresponding parameters that are required for each TCL scripts.

Table 10 TCL Scripts and Parameters

TCL Script Name	Description —Summary	Parameters to Configure
clid_4digits_npw_3_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. The length of digits allowed for the account number and password are configurable through the command-line interface (CLI). If the authentication fails, it allows the caller to retry. The retry number is also configured through the CLI.	• **call application voice uid-len** min = 1, max = 20, default =10 • **call application voice pin-len** min = 0, max = 10, default = 4 • **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_col_npw_cli.tcl	This script authenticates the account number and PIN respectively using ANI and NULL. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_authen_collect_cli.tcl	This script authenticates the account number and PIN using ANI and DNIS. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected separately.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_3_cli.tcl	This script authenticates using ANI and NULL for account and PIN respectively. If the authentication fails, it allows the caller to retry. The retry number is configured through the CLI.	• **call application voice retry-count** min = 1, max = 5, default = 3
clid_col_npw_npw_cli.tcl	This script authenticates using ANI and NULL for account and pin respectively. If authentication fails, it allows the caller to retry. The retry number is configured through the CLI. The account number and PIN are collected together.	• **call application voice retry-count** min = 1, max = 5, default = 3

Examples

The following example shows how to configure a 30-second warning time for the application named prepaid:

```
call application voice prepaid warning-time 30
```

Related Commands

Command	Description
call application voice language	Defines the language of the audio file for the designated application and passes that information to the application.
call application voice load	Reload this designated TCL script.
call application voice location	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice pin-len	Defines the number of characters in the PIN for the application and passes that information to the application.
call application voice redirect-number	Defines the telephone number to which a call will be redirected—for example, the operator telephone number of the service provider—for the designated application.
call application voice retry-count	Defines the number of times a caller is permitted to reenter the PIN for a designated application and passes that information to the application.
call application voice set-location	Defines the location, language, and category of the audio files for the designated application and passes that information to the application.
call application voice uid-len	Defines the number of characters in the UID for the designated application and passes that information to the application..

call-waiting

To enable call waiting, use the **call-waiting** command in interface configuration mode. To disable call waiting, use the **no** form of this command.

call-waiting

no call-waiting

Syntax Description

This command has no arguments or keywords.

Defaults

Call waiting is enabled.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command is applicable for Cisco 800 series routers.

You must specify this command when creating a dial peer. This command will not work if it is not specified within the context of a dial peer. For information on creating a dial peer, refer to the *Cisco 800 Series Routers Software Configuration Guide*.

Examples

The following example disables call waiting:

```
no call-waiting
```

Related Commands

Command	Description
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer."
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
port (dial-peer)	Enables an interface on a PA-4R-DTR port adapter to operate as a concentrator port.
ring	Sets up a distinctive ring for telephones, fax machines, or modems connected to a Cisco 800 series router.
show dial-peer voice	Displays configuration information and call statistics for dial peers.

called-number (dial-peer)

To enable an incoming Voice over Frame Relay call leg to get bridged to the correct POTS call leg when a static FRF.11 trunk connection is used, use the **called-numbered** command in dial-peer configuration mode. To disable a static trunk connection, use the **no** form of this command.

called-number *string*

no called-number

Syntax Description

string	A string of digits including wildcards that specifies the telephone number of the voice-port dial peer

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

This command applies to the Cisco 2600 series and 3600 series routers only. It is ignored on the Cisco MC3810 and the Cisco 7200 series.

The **called-number** (dial-peer) command is used only when the dial peer type is VoFR and you are using the frf11-trunk (FRF.11) session protocol; it is ignored at all times on the Cisco MC3810, and on all other platforms when using the cisco-switched session protocol.

Because FRF.11 does not provide any end-to-end messaging to manage a trunk, the **called-number** (dial-peer) command is necessary to allow the router to establish an incoming trunk connection. The E.164 number is used to find a matching dial peer during call setup.

Examples

The following example shows how to configure a Cisco 2600 series or 3600 series router for a static FRF.11 trunk connection to a specific telephone number (555-2150), beginning in global configuration mode:

```
voice-port 1/0/0
 connection trunk 5558000
 exit

dial-peer voice 100 pots
 destination pattern 5552150
 exit

dial-peer voice 200 vofr
 session protocol frf11-trunk
 called-number 5552150
 destination pattern 5558000
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
connection	Specifies a connection mode for a voice port.
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.
fax-rate	Establishes the rate at which a fax will be sent to the specified dial peer.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
signal-type	Sets the signalling type to be used when connecting to a dial peer.
vad (dial peer)	Enables VAD for the calls using a particular dial peer.

cap-list vfc

To add a voice codec overlay file to the capability file list, use the **cap-list vfc** command in global configuration mode. To disable a particular codec overlay file that has been added to the capability list, use the **no** form of this command.

cap-list *filename* **vfc** *slot-number*

no cap-list *filename* **vfc** *slot-number*

Syntax Description

filename	Identifies the codec file stored in VFC Flash memory.
slot-number	Identifies the slot where the VFC is installed. Valid values are 0, 1, and 2.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

When VCWare is unbundled, it automatically adds DSPWare to Flash memory, creates both the capability and default file lists, and populates these lists with the default files for that version of VCWare. The capability list defines the available voice codecs for H.323 capability negotiation. Use the **cap-list vfc** command to add the indicated voice codec overlay file (defined by *filename*) to the capability file list in Flash memory.

Examples

The following example adds the following codec to the list included in Flash memory:

```
config terminal
 cap-list cdc-g711-1.0.14.0.bin vfc 0
```

Related Commands

Command	Description
default-file vfc	Specifies an additional (or different) file from the ones in the default file list and stored in VFC Flash memory.

card type

To configure the card type on the port adapter of the Cisco 7200 series router, use the **card type** command in global configuration mode. Use the **no** form of this command to restore the default value.

card type {**t1** | **e1**} *slot* [*bay*]

no card type

Syntax Description

t1	Specifies T1connectivity of 1.544 Mbps through the telephone-switching network, using AMI or B8ZS coding.
e1	Specifies wide-area digital transmission scheme used predominately in Europe. that carries data at a rate of 2.048 Mbps.
slot	Slot number of the interface.
bay	(Optional) Card interface bay number in a slot (RSP platform only).

Defaults

No default behavior or values.

Command Modes

Global configuration

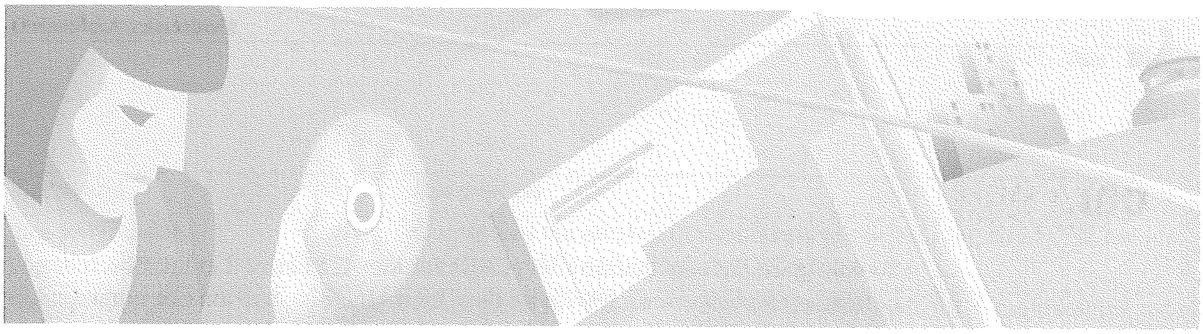
Command History

12.0(5)XE and 12.0(7)T	This command was introduced.

Examples

The following example configures T1 data transmission on port 1 on the Cisco 7200 series router:

```
card type t1 1
```



Multiservice Applications Commands: Cb through D

This book documents commands used to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features. Commands in this book are listed alphabetically. For information on how to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features, refer to the *Cisco IOS Multiservice Applications Configuration Guide*.

cbr

To configure the constant bit rate (CBR) for the ATM circuit emulation service (CES) for an ATM permanent virtual circuit (PVC) on the Cisco MC3810, use the **cbr** command in ATM virtual circuit configuration mode. To restore the default, use the **no** form of this command.

cbr *rate*

no cbr *rate*

Syntax Description

rate	Constant bit bate (also known as the average cell rate) for ATM CES. The valid range for this command is from 56 to 10,000 kbps.

Defaults

0

Command Modes

ATM virtual circuit configuration

Command History

Release	Modification
12.0	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

Examples

The following example configures the constant bit rate on ATM PVC 20 on the Cisco MC3810:

```
pvc 20
 cbr 56
```

Related Commands

Command	Description
ces cell-loss-integration-period	Sets the CES cell-loss integration period on the Cisco MC3810 multiservice concentrator.
ces clockmode synchronous	Configures the ATM CES synchronous clock mode on the Cisco MC3810 multiservice concentrator.
ces connect	Maps the CES service to an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces initial-delay	Configures the size of the receive buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.
ces max-buf-size	Configures the send buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.
ces max-buf-size	Configures the send buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.

Command	Description
ces partial-fill	Configures the number of user octets per cell for the ATM CES on the Cisco MC3810 multiservice concentrator.
ces service	Configures the ATM CES type on the Cisco MC3810 multiservice concentrator.
encapsulation atm-ces	Enables CES ATM encapsulation on the Cisco MC3810 multiservice concentrator.

ccs connect

To configure a CCS connection on an interface configured to support CCS frame forwarding, use the **ccs connect** command in controller interface configuration mode. To disable the CCS connection on the interface, use the **no** form of this command.

ccs connect {**serial** | **atm**} *number* [**dlci** *dlci* | **pvc** *vci* | **pvc** *vcd* | **pvc** *vpi/vci* | **pvc** *string*]

no ccs connect {**serial** | **atm**} *number* [**dlci** *dlci* | **pvc** *vci* | **pvc** *vcd* | **pvc** *vpi/vci* | **pvc** *string*]

Syntax Description

serial	Make a serial CCS connection.
atm	Make an ATM CCS connection.
number	Specifies the connection number.
dlci *dlci*	(Optional) Specifies the DLCI number.
pvc *vci*	(Optional) Specifies the PVC virtual circuit identifier.
pvc *vcd*	(Optional) Specifies the PVC virtual circuit descriptor.
pvc *vpi/vci*	(Optional) Specifies the PVC virtual path identifier/virtual channel identifier.
pvc *string*	(Optional) Specifies the PVC string.

Defaults

No CCS connection is made.

Command Modes

Controller interface configuration

Command History

Release	Modification
12.0(2)T	This command was introduced.

Examples

The following example shows how to configure a CCS frame-forwarding connection on DLCI 100:

```
ccs connect serial 1 dlci 100
```

The following example shows how to configure a CCS frame-forwarding connection over an ATM PVC:

```
ccs connect atm0 pvc 100
```

Related Commands

Command	Description
mode ccs	Configures the T1/E1 controller to support CCS cross-connect or CCS frame-forwarding.

ces cell-loss-integration-period

To set the circuit emulation service (CES) cell-loss integration period, use the **ces cell-loss-integration-period** command in interface configuration mode.To delete the cell-loss integration period, use the **no** form of this command.

ces cell-loss-integration-period *period*

no ces cell-loss-integration-period *period*

Syntax Description

period	Time in milliseconds for the cell loss integration period. Possible values are from 1 to MAXINT.

Defaults

2500

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

This command is supported on serial ports 0 and 1 with **encapsulation atm-ces**.

Examples

The following example configures the CES cell-loss integration period on serial port 0 to 1056:

```
interface serial 0
 ces cell-loss-integration-period 1056
```

Related Commands

Command	Description
cbr	Configures the CBR for the ATM CES for an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces clockmode synchronous	Configures the ATM CES synchronous clock mode on the Cisco MC3810 multiservice concentrator.
ces connect	Maps the CES service to an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces initial-delay	Configures the size of the receive buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.
ces max-buf-size	Configures the send buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.

Command	Description
ces partial-fill	Configures the number of user octets per cell for the ATM CES on the Cisco MC3810 multiservice concentrator.
ces service	Configures the ATM CES type on the Cisco MC3810 multiservice concentrator.
encapsulation atm-ces	Enables CES ATM encapsulation on the Cisco MC3810 multiservice concentrator.

ces clockmode synchronous

To configure the ATM circuit emulation service (CES) synchronous clock mode, use the **ces clockmode synchronous** command in interface configuration mode. To restore the default value, use the **no** form of this command.

ces clockmode synchronous

no ces clockmode synchronous

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

This command maps into the transmit clock source of the CBR interface. This command is supported on serial ports 0 and 1 when set for CES ATM encapsulation.

Examples

The following example sets the ATM CES clock to synchronous mode on serial port 0:

```
interface serial 0
 ces clockmode synchronous
```

Related Commands

Command	Description
encapsulation atm-ces	Enables CES ATM encapsulation on the Cisco MC3810 multiservice concentrator.

ces connect

To map the circuit emulation service (CES) service to an ATM PVC on the Cisco MC3810, use the **ces connect** command in interface configuration mode. To delete the CES map to the ATM PVC, use the **no** form of this command.

ces connect *atm-interface* **pvc** [*name* | [*vpi/*]*vci*]

no ces connect *atm-interface* **pvc** [*name* | [*vpi/*]*vci*]

Syntax Description

atm-interface	Number of the ATM interface. The only valid option on the Cisco MC3810 is ATM0.
pvc	Specifies that the connection is to an ATM PVC.
name	(Optional) The name of the ATM PVC.
vpi/	(Optional) The virtual path identifier value.
vci	(Optional) The virtual channel identifier value.

Defaults

No ATM interface is defined.

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

This command is supported on serial ports 0 and 1. The ATM interface must be configured to **encapsulation atm-ces**, and the vpi/vci must be defined on the interface.

Examples

The following example maps the CES service to PVC 20 on ATM port 0:

```
ces connect atm0 pvc 20
```

Related Commands

Command	Description
cbr	Configures the CBR for the ATM CES for an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces cell-loss-integration-period	Sets the CES cell-loss integration period on the Cisco MC3810 multiservice concentrator.
ces clockmode synchronous	Configures the ATM CES synchronous clock mode on the Cisco MC3810 multiservice concentrator.

Command	Description
ces initial-delay	Configures the size of the receive buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.
ces max-buf-size	Configures the send buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.
ces partial-fill	Configures the number of user octets per cell for the ATM CES on the Cisco MC3810 multiservice concentrator.
ces service	Configures the ATM CES type on the Cisco MC3810 multiservice concentrator.
encapsulation atm-ces	Enables CES ATM encapsulation on the Cisco MC3810 multiservice concentrator.

ces initial-delay

To configure the size of the receive buffer of a circuit emulation service (CES) circuit, use the **ces initial-delay** command in interface configuration mode. To remove the initial-delay value, use the **no** form of this command.

ces initial-delay *bytes*

no ces initial-delay *bytes*

Syntax Description

bytes	The size of the receive buffer of the CES circuit. The valid range is from 1 to 16,000 bytes. This command is used to accommodate cell jitter on the network. Bytes received from the ATM network are buffered by this amount before being sent to the CES port.

Defaults

4000 bytes

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

Examples

The following example configures the transmit buffer of the CES circuit to 8000 bytes:

```
ces initial-delay 8000
```

Related Commands

Command	Description
ces cell-loss-integration-period	Sets the CES cell-loss integration period on the Cisco MC3810 multiservice concentrator.
ces clockmode synchronous	Configures the ATM CES synchronous clock mode on the Cisco MC3810 multiservice concentrator.
ces connect	Maps the CES service to an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces max-buf-size	Configures the send buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.

Command	Description
ces partial-fill	Configures the number of user octets per cell for the ATM CES on the Cisco MC3810 multiservice concentrator.
ces service	Configures the ATM CES type on the Cisco MC3810 multiservice concentrator.

ces max-buf-size

To configure the transmit buffer of a circuit emulation service (CES) circuit, use the **ces max-buf-size** command in interface configuration mode. To delete the CES transmit buffer size, use the **no** form of this command.

ces max-buf-size *size*

no ces max-buf-size *size*

Syntax Description

size	Maximum size of the transmit buffer for the CES. Possible values are from 80 to 1520.

Defaults

256

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

Using this command, incoming bytes received on a CES port are buffered by the amount configured, and sent to the AAL1 process as a block of data.

This command is supported on serial ports 0 and 1 when the **encapsulation atm-ces** command is enabled.

Examples

The following example configures the maximum CES reassembly buffer size to 1520:

```
ces max-buf-size 1520
```

Related Commands

Command	Description
ces cell-loss-integration-period	Sets the CES cell-loss integration period on the Cisco MC3810 multiservice concentrator.
ces clockmode synchronous	Configures the ATM CES synchronous clock mode on the Cisco MC3810 multiservice concentrator.
ces connect	Maps the CES service to an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces initial-delay	Configures the size of the receive buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.

Command	Description
ces partial-fill	Configures the number of user octets per cell for the ATM CES on the Cisco MC3810 multiservice concentrator.
ces service	Configures the ATM CES type on the Cisco MC3810 multiservice concentrator.

ces partial-fill

To configure the number of user octets per cell for the ATM circuit emulation service (CES), use the **ces partial-fill** command in interface configuration mode. To delete the CES partial-fill value, use the **no** form of this command.

ces partial-fill *octet*

no ces partial-fill *octet*

Syntax Description

octet	Number of user octets per cell for the CES. Possible values of octet range from 0 to 47. Setting this number to zero disables partial cell fill and causes all cells to be completely filled before they are sent.

Defaults

47

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

Setting the value of the **ces partial-fill** command to zero disables partial cell fill and causes all cells to be completely filled before they are sent. This command is supported on serial ports 0 and 1 when the **encapsulation atm-ces** command is enabled.

Examples

The following example sets the CES partial cell fill to 20 octets per cell for serial port 0:

```
interface serial 0
 ces partial-fill 20
```

Related Commands

Command	Description
ces cell-loss-integration-period	Sets the CES cell-loss integration period on the Cisco MC3810 multiservice concentrator.
ces clockmode synchronous	Configures the ATM CES synchronous clock mode on the Cisco MC3810 multiservice concentrator.
ces connect	Maps the CES service to an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces initial-delay	Configures the size of the receive buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.

Command	Description
ces max-buf-size	Configures the send buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.
ces service	Configures the ATM CES type on the Cisco MC3810 multiservice concentrator.

ces service

To configure the ATM circuit emulation service (CES) type, use the **ces service** command in interface configuration mode. To stop the ATM CES service type, use the **no** form of this command.

ces service structured

no ces service structured

Syntax Description

structured	Specifies that the ATM CES type is structured. Structured is the only option supported in this release.

Defaults

Structured

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

This command is supported on serial ports 0 and 1 when the **encapsulation atm-ces** command is enabled.

Examples

The following example sets the CES service to structured for serial port 0:

```
interface serial 0
 ces service structured
```

Related Commands

Command	Description
ces cell-loss-integration-period	Sets the CES cell-loss integration period on the Cisco MC3810 multiservice concentrator.
ces clockmode synchronous	Configures the ATM CES synchronous clock mode on the Cisco MC3810 multiservice concentrator.
ces connect	Maps the CES service to an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces initial-delay	Configures the size of the receive buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.

Command	Description
ces max-buf-size	Configures the send buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.
ces partial-fill	Configures the number of user octets per cell for the ATM CES on the Cisco MC3810 multiservice concentrator.

clear cable flap-list

To reset the flap-list table for a specific cable modem or for all cable modems connected to the Cisco uBR7200 series, use the **clear cable flap-list** command in privileged EXEC mode.

clear cable flap-list [*mac-addr* | **all**]

Syntax Description

mac-addr	(Optional) MAC address. Specify the 48-bit hardware address of an individual cable modem.
all	(Optional) Remove all cable modems from the flap-list table.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Cable modems are removed from the flap-list table after the number of days (between 1 and 60) specified by the **cable flap-list aging** global configuration command. Use the **clear cable flap-list** command to remove individual cable modems from the flap-list while retaining flapping activity for other cable modems, or to clear the entire flap-list table.

Examples

The following example removes all the cable modems from the flap-list table:

```
clear cable flap-list all
```

Related Commands

Command	Description
cable flap-list aging	Specifies how many days to record and retain flapping activity on a cable modem before aging the cable modem out of the flap-list table.
cable flap-list insertion-time	Sets the insertion time interval that determines whether a cable modem is placed in the flap list.
cable flap-list power-adjust threshold	Specifies the power-adjust threshold for recording a cable modem flap-list event.
cable flap-list size	Specifies the maximum number of cable modems reported in the flap-list table.

clear cable modem counters

To reset the cable modem flapping counters to zero, use the **clear cable modem counters** command in privileged EXEC mode.

clear cable modem {*mac-addr* | *ip-addr* | **all**} **counters**

Syntax Description

mac-addr	MAC address. Specify the 48-bit hardware address of an individual cable modem.
ip-addr	IP address. Specify the IP address of an individual cable modem.
all	Resets the flapping data for all modems.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following example clears the counters for the cable modem at IP address 172.00.00.00:

```
clear cable modem 172.00.00.00 counters
```

Related Commands

Command	Description
clear cable modem reset	Removes a cable modem from the Station Maintenance List and resets the cable modem.

clear cable modem reset

To remove a cable modem from the Station Maintenance List and reset it, use the **clear cable modem reset** command in privileged EXEC mode.

clear cable modem {*mac-addr* | *ip-addr* | **all**} **reset**

Syntax Description

mac-addr	MAC address. Specify the 48-bit hardware address of an individual cable modem.
ip-addr	IP address. Specify the IP address of an individual cable modem.
all	Removes all the cable modems from the Station Maintenance List.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

This command causes the link to the cable modem to drop. The cable modem responds by resetting itself. It can take up to 30 seconds for the reset sequence to begin.

Examples

The following example removes the cable modem at 172.00.00.00 from the Station Maintenance List and causes it to reset:

```
clear cable modem 172.00.00.00 reset
```

Related Commands

Command	Description
clear cable modem counters	Resets the flapping counters of a cable modem to zero.

clear csm-statistics modem

To clear the CSM statistics for a modem or group of modems, use the **clear csm-statistics modem** command in privileged EXEC mode.

clear csm-statistics modem [*slot/port* | *modem-group-number*]

Syntax Description

slot/port	(Optional) Identifies the location (and thereby the identity) of a specific modem.
modem-group-number	(Optional) Designates a defined modem group.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use the **clear csm-statistics modem** command to clear CSM statistics for a particular modem or group of modems. If the *slot/port* argument is specified, the CSM call statistics for calls using the identified modem will be cleared. If a modem group number is specified, then the CSM call statistics for calls using the modems associated with that group will be cleared. If no argument is specified, all CSM call statistics for all modems will be cleared.

Examples

The following example clears CSM call statistics for calls coming in on modems associated with modem group 2:

```
config terminal
 clear csm-statistics modem 2
```

Related Commands

Command	Description
clear csm-statistics voice	Clears the CSM statistics for a particular or all DSP channels.

clear csm-statistics voice

To clear the CSM statistics for a particular or all digital signal processor (DSP) channels, use the **clear csm-statistics voice** command in privileged EXEC mode.

clear csm-statistics voice [*slot/dspm/dsp/dsp-channel*]

Syntax Description

slot/dspm/dsp/dsp-channel	(Optional) Identifies the location of a particular DSP channel.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

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

Examples

The following example clears CSM call statistics for calls coming in on all DSP channels:

```
config terminal
 clear csm-statistics voice
```

Related Commands

Command	Description
clear csm-statistics modem	Clears the CSM statistics for a modem or group of modems.

clear h323 gatekeeper call

To force a disconnect on a specific call or all calls active on a particular Multimedia Conference Manager (MCM) gateway, use the **clear h323 gatekeeper call** command in privileged EXEC mode.

clear h323 gatekeeper call {**all** | **local-callID** *local-callID*}

Syntax Description

all	Forces all active calls currently associated with this MCM gatekeeper to be disconnected.
local-callID	Forces a single active call associated with this MCM gatekeeper to be disconnected.
local-callID	The local call identification number (CallID) that identifies the call to be disconnected.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)T	This command was introduced.

Usage Guidelines

If you want to force a particular call to be disconnected (as opposed to all active calls on the MCM gateway), use the CallID number to identify that specific call. You can find the local CallID number for a specific call by using the **show gatekeeper calls** command; the ID number is displayed in the LocalCallID column. Figure 1 shows output from the **show gatekeeper calls** command.

Figure 1 show gatekeeper calls Command Output

```
router# show gatekeeper calls

Total number of active calls =1
                    Gatekeeper Call Info
                    ====================
LocalCallID                   Age (secs)       BW
12-3339                       94               768 (Kbps)
 Endpt(s): Alias     E.164Addr      CallSignalAddr   Port    RASSignalAddr    Port
   src EP: epA                      10.0.0.11        1720    10.0.0.11        1700
   dst EP: epB2zoneB.com
   src PX: pxA                      10.0.0.1         1720    10.0.0.11        24999
   dst PX: pxB                      172.21.139.90    1720    172.21.139.90    24999
```

Examples

The following example forces an active call on the MCM gateway to be disconnected. The local ID number of the active call is 12-3339.

```
enable
clear h323 gatekeeper call local-callID 12-3339
```

The following example forces all active calls on the MCM gateway to be disconnected:

```
enable
clear h323 gatekeeper call all
```

Related Commands

Command	Description
show gatekeeper calls	Shows the status of each ongoing call that a gatekeeper is aware of.

clear voice port

To clear voice port calls in progress on the Cisco MC3810, use the **clear voice port** command in privileged EXEC mode.

clear voice port [*slot/port*]

Syntax Description

slot/port	(Optional) The voice port slot number and port number. If you do not specify a voice port, all calls on all voice ports are cleared.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

If you do not specify a voice port, all calls on all voice ports are cleared. A confirmation prompt is displayed.

Examples

The following example clears all calls on voice port 1/2 on the Cisco MC3810:

```
clear voice port 1/2
```

clock rate line

To configure the line clock rate for serial ports 0 or 1 in DTE mode on the Cisco MC3810, use the **clock rate line** command in interface configuration mode. To cancel the clock rate line value, use the **no** form of this command.

clock rate line *rate*

no clock rate line *rate*

Syntax Description

rate	Network clock line rate in bits per second. The range is from 56 kbps to 2048 kbps. The value entered should be a multiple of 8,000 of the value set for the **network-clock base-rate** command. There is no default rate.

Defaults

No clock rate is set.

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command specifies the rate of the incoming clock so that the appropriate internal clock scaling can be performed.

To configure the clock rate for a serial port in DTE mode, use the **clock rate network-clock** command.

Examples

The following configures the clock rate on serial 1 in DTE mode:

```
interface serial 1
 clock rate line 2048
```

Related Commands

Command	Description
clock rate network-clock	Configures the network clock speed for serial ports 0 or 1 in DCE mode.
clock source (MC3810 multiservice concentrator)	Specifies the clock source of a DS1 link on the Cisco MC3810 multiservice concentrator.
network-clock base-rate	Configures the network clock base rate for universal I/O serial ports 0 and 1 on the Cisco MC3810 multiservice concentrator.

clock rate network-clock

To configure the network clock speed for serial ports 0 or 1 in DCE mode on the Cisco MC3810, use the **clock rate network-clock** command in interface configuration mode. To cancel the network clock speed value, use the **no** form of this command.

clock rate network-clock *rate*

no clock rate network-clock *rate*

Syntax Description

rate	Network clock speed in bits per second. The range is from 56 kbps to 2048 kbps. The value entered should be a multiple of the value set for the **network-clock base-rate** command. There is no default rate.

Defaults

No clock rate is set.

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command uses a synchronized clock on the serial port. The use of this command allows the clock on the serial port to be synchronized with the clock source of controller T1 0.

To configure the clock rate for a serial port in DTE mode, use the **clock rate line** command.

Examples

The following configures the clock rate on serial 1 in DCE mode:

```
interface serial 1
 clock rate network-clock 2048
```

Related Commands

Command	Description
clock rate line	Configures the line clock rate for serial ports 0 or 1 in DTE mode.
clock source (MC3810 multiservice concentrator)	Specifies the clock source of a DS1 link on the Cisco MC3810 multiservice concentrator.
network-clock base-rate	Configures the network clock base rate for universal I/O serial ports 0 and 1 on the Cisco MC3810 multiservice concentrator.

codec (Cisco 7200 series)

To specify call density and codec complexity based on a particular codec standard, use the **codec** command in DSP interface dsp farm mode. To reset the card type to the default, use the **no** form of the command.

codec {high | low | medium}

no codec {high | low | medium}

Syntax Description

high	Specifies high complexity: Two channels of any mix of codec.
low	Specifies low complexity: Eight channels of g711.
medium	Specifies medium complexity: Four channels of g711/g726/g729a/fax.

Defaults

Medium

Command Modes

DSP interface dsp farm

Command History

Release	Modification
12.0(5)XE and 12.0(7)T	This command was introduced.

Usage Guidelines

Codec complexity refers to the amount of processing required in order to perform compression. Codec complexity affects the number of calls that can take place on the DSPfarm interfaces, referred to as call density. The greater the codec complexity, the fewer calls are handled. For example, G.711 requires less DSP processing than G.728, so that as long as the bandwidth is available, more calls can be handled simultaneously by using the G.711 standard than using G.728.

The DSP interface dspfarm **codec** complexity setting affects the options available for the **codec** dial-peer configuration command.

To change codec complexity, you must first remove any configured CAS or DS0 groups, and then reinstate them after the change.

Examples

The following example configures the DSPfarm interface 1/0 on the Cisco 7200 series routers to support high compression:

```
dspint dspfarm 1/0
 codec high 0-30
```

Related Commands

Command	Description
command-type	Specifies the companding standard used to convert between analog and digital signals in PCM systems.

codec (dial-peer)

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

Cisco 2600 and 3600 series routers, Cisco AS5300 access servers, and AS5800 access servers

codec {**g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g728** | **g729br8** | **g729r8**} [*bytes* | **bytes** *payload_size*] [**pre-ietf**]

no codec {**g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g728** | **g729br8** | **g729r8**} [*bytes* | **bytes** *payload_size*] [**pre-ietf**]

Cisco MC3810

codec {**g711alaw** | **g711ulaw** | **g726r32** | **g729ar8** | **g729r8**} [*bytes* | **bytes** *payload_size*]

no codec {**g711alaw** | **g711ulaw** | **g726r32** | **g729ar8** | **g729r8**} [*bytes* | **bytes** *payload_size*]

Syntax Description

Cisco 2600 and 3600 series routers, Cisco AS5300 access servers, and AS5800 access servers

g711alaw	G.711 a-Law at 64000 bits per second (bps).
g711ulaw	G.711 u-Law at 64000 bps.
g723ar53	G.723.1 ANNEX A at 5300 bps.
g723ar63	G.723.1 ANNEX A at 6300 bps.
g723r53	G.723.1 at 5300 bps.
g723r63	G.723.1 at 6300 bps.
g726r16	G.726 at 16000 bps.
g726r24	G.726 at 24000 bps.
g726r32	G.726 at 32000 bps.
g728	G.728 at 16000 bps.
g729br8	G.729 ANNEX B at 8000 bps.
g729r8	G.729 at 8000 bps. This is the default codec.
bytes	(Optional) Specifies the voice data bytes per frame for VoIP dial peers. Acceptable values are from 10 to 240 in increments of 10 (10, 20, 30 ... 220, 230, 240). Any other value is rounded down.
bytes	(Optional) Used to specify the number of bytes in the voice payload of each VoFR frame of a VoFR dial peer.
payload_size	(Optional) The number of bytes in the voice payload of each VoFR frame. Enter a **?** character after the keyword **bytes** to get a list of valid payload values for your specific VoFR dial peer.
pre-ietf	(Optional) Specifies pre-Internet Engineering Task Force (IETF) bit-ordering. This keyword is valid only on the Cisco 2600, 3600, or AS5300 routers when the **g729r8** codec is specified. You *must* specify this keyword for connection to a Cisco 2600 series, 3600 series, 7200 series router, or AS5300 access server running a Cisco IOS release prior to 12.0(5)T or 12.0(4)XH.

Cisco MC3810

g711alaw	G.711 a-Law at 64,000 bits per second (bps).
g711ulaw	G.711 u-Law at 64,000 bps.
g726r32	G.726 at 32000 bps.
g729ar8	G.729 ANNEX A at 8000 bps.
g729r8	G.729 at 8000 bps. This is the default codec.
bytes	(Optional) Specifies the voice data bytes per frame for VoIP dial peers. Acceptable values are from 10 to 240 in increments of 10 (10, 20, 30 ... 220, 230, 240). Any other value is rounded down.
bytes	(Optional) Used to specify the number of bytes in the voice payload of each VoFR frame of a VoFR dial peer.
payload_size	(Optional) The number of bytes in the voice payload of each VoFR frame. Enter a **?** character after the keyword **bytes** to get a list of valid payload values for your specific VoFR dial peer.

Defaults

g729r8, 30-byte payload for VoFR, VoATM, and VoHDLC

g729r8, 20-byte payload for VoIP

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(4)T	This command was modified for VoFR dial peers, and support for this command was extended to the Cisco MC3810.
12.0(5)XK and 12.0(7)T	Additional codec choices and other options were added.

Usage Guidelines

Use this command to define a specific voice coder rate of speech and payload size for a VoIP dial peer or for a VoFR dial peer.

A specific codec type can be configured on the dial-peer as long as it is supported by the setting used with the **codec complexity** voice-card configuration command. The **codec complexity** command is voice-card-specific and platform-specific.

The **codec** dial-peer configuration command is particularly useful when you must change to a small-bandwidth codec. Large-bandwidth codecs, such as G.711, do not fit in a small-bandwidth link. However, the **g711alaw** and **g711ulaw** codecs provide higher-quality voice transmission than other codecs. The **g729r8** codec provides near-toll quality with considerable bandwidth savings.

If codec values for the dial peers of a connection do not match, the call fails.

You can change the payload of each VoIP frame by using the *bytes* setting; you can change the payload of each VoFR frame by using the **bytes** keyword with the *payload_size* setting. However, increasing the payload size can add processing delay for each voice packet.

For toll quality, use the **g711alaw** or **g711ulaw** keyword. 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** keyword.

On the Cisco MC3810, this command was first supported as a voice port command. This command does not apply to the Cisco 7200 series routers.

On the Cisco MC3810, you can also assign codec values to the voice port. If configuring calls to a Cisco MC3810 running software versions prior to 12.0(4)T, configure the **codec** command on the voice port. If configuring Cisco-trunk permanent calls, configure the **codec** command on the dial peer. If you configure the **codec** command on the dial peer for Voice over Frame Relay permanent calls on the Cisco MC3810, the dial peer **codec** command setting overrides the **codec** setting configured on the voice port.

Note For regular switched calls on the Cisco MC3810, the codec value must be configured on the voice port, and the voice payload size is not configurable.

Examples

The following example shows how to configure a voice coder rate that provides toll quality voice with a payload of 120 bytes per voice frame on a Cisco 2600 series or 3600 series router acting as a terminating node. The example configuration begins in global configuration mode and is for VoFR dial peer 200.

```
dial-peer voice 200 vofr
 codec g711alaw bytes 120
```

The following example configures a voice coder rate for VoIP dial peer 10 that provides toll quality but uses a relatively high amount of bandwidth:

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

Related Commands

Command	Description
called-number (dial-peer)	Enables an incoming VoFR call leg to get bridged to the correct POTS call leg when using a static FRF.11 trunk connection.
codec complexity	Sets codec complexity and call density for voice cards.
connection	Specifies a connection mode for a voice port.
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.
fax-rate	Establishes the rate at which a fax will be sent to the specified dial peer.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.

Command	Description
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
show dial-peer voice	Displays the codec setting for dial peers.
vad (dial peer)	Enables VAD for the calls using a particular dial peer.

codec (voice-port)

To configure voice compression on the Cisco MC3810 voice port, use the **codec** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

codec {**g729r8** | **g729ar8** | **g726r32** | **g711alaw** | **g711ulaw**}

no codec {**g729r8** | **g729ar8** | **g726r32** | **g711alaw** | **g711ulaw**}

Syntax Description

g729r8	Specifies G729, 8k CSA-CELP compression. This is the default.
g729ar8	Specifies G729, 8k CSA-CELP Annex A compression.
g726r32	Specifies G.726 32K ADCPM compression.
g711alaw	Specifies G.711 64K PCM a-Law compression.
g711ulaw	Specifies G.711 64K PCM u-Law compression.

Defaults

The default is **g729ar8** compression mode.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

The **g729ar8** compression mode can support a maximum of 24 simultaneously active on-net voice calls on the Cisco MC3810 while the **g729r8** compression mode can only support a maximum of 12. Both compression modes have a nominal data rate of 8 kbps.

This command applies to both analog and digital voice ports on the Cisco MC3810.

Note On the Cisco 3600 series, the codec compression values are assigned to the dial peer using the **codec** dial-peer configuration command.

Examples

The following example configures voice port 1/1 on the Cisco MC3810 to support g729r8 compression:

```
voice-port 1/1
 codec g729r8
```

Related Commands

Command	Description
compand-type	Specifies the companding standard used to convert between analog and digital signals in PCM systems on the Cisco MC3810 multiservice concentrator.

codec complexity

To specify call density and codec complexity based on the codec standard you are using, use the **codec complexity** command in voice-card configuration mode. To reset the voice card to the default, use the **no** form of the command.

codec complexity {**high** | **medium**}

no codec complexity {**high** | **medium**}

Syntax Description

high	High-complexity codecs support the following services: G.711, G.726, G.729, G.729 Annex B, G.723.1, G.723.1 Annex A, G.728, and fax relay.
medium	Medium-complexity codecs support the following services: G.711, G.726, G.729 Annex A, G.729 Annex B with Annex A, and fax relay.

Defaults

medium

Command Modes

Voice-card configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	The command was introduced for the Cisco 2600 and 3600 series.

Usage Guidelines

Codec complexity refers to the amount of processing required in order to perform compression. Codec complexity affects the number of calls that can take place on a voice card's digital signal processors (DSPs), referred to as call density. The greater the codec complexity, the fewer calls are handled. For example, G.711 requires less DSP processing than G.728, so that as long as the bandwidth is available, more calls can be handled simultaneously by using the G.711 standard than using G.728.

All voice cards in a router must use the same codec complexity. The voice-card **codec complexity** setting affects the options available for the **codec** dial-peer configuration command.

To change codec complexity, you must first remove any configured CAS or DS0 groups, then reinstate them after the change.

If you set codec complexity to **high**, the following options are available:

- **g711alaw**—G.711 a-Law 64,000 bps
- **g711ulaw**—G.711 u-Law 64,000 bps
- **g723ar53**—G.723.1 Annex A 5300 bps
- **g723ar63**—G.723.1 Annex A 6300 bps
- **g723r53**—G.723.1 5300 bps
- **g723r63**—G.723.1 6300 bps
- **g726r16**—G.726 16,000 bps

- **g726r24**—G.726 24,000 bps
- **g726r32**—G.726 32,000 bps
- **g728**—G.728 16,000 bps
- **g729r8**—G.729 8000 bps (default)
- **g729br8**—G.729 Annex B 8000 bps

If you set codec complexity to **medium**, the following options are valid:

- **g711alaw**—G.711 a-Law 64,000 bps
- **g711ulaw**—G.711 u-Law 64,000 bps
- **g726r16**—G.726 16,000 bps
- **g726r24**—G.726 24,000 bps
- **g726r32**—G.726 32,000 bps
- **g729r8**—G.729 Annex A 8000 bps
- **g729br8**—G.729 Annex B with Annex A 8000 bps

Examples

The following example configures a voice card for high-complexity codecs:

```
voice-card 1
 codec complexity high
```

Related Commands

Command	Description
ds0-group	Defines T1/E1 channels for compressed voice calls and the CAS method by which the router connects to the PBX or PSTN.

codec preference

To specify a list of preferred codecs to use on a dial peer, use the **codec preference** command in class configuration mode. To restore the default, use the **no** form of this command.

codec preference *value codec_type* **bytes** *size*

no codec preference *value codec_type* **bytes** *size*

Syntax Description

value	Specifies the order of preference with 1 being the most preferred and 12 being the least preferred.
codec_type	Specifies the type of codec preferred.
bytes	Specifies that the size of the voice frame is in bytes.
size	Number of voice data bytes per frame. Valid sizes vary by codec.

Defaults

No default behavior or values.

Command Modes

Class configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced.

Examples

The following example illustrates a list of 12 codecs in order of preference:

```
voice class codec 99
 codec preference 1 g711alaw
 codec preference 2 g711ulaw bytes 80
 codec preference 3 g723ar53
 codec preference 4 g723ar63 bytes 144
 codec preference 5 g723r53
 codec preference 6 g723r63 bytes 120
 codec preference 7 g726r16
 codec preference 8 g726r24
 codec preference 9 g726r32 bytes 80
 codec preference 10 g728
 codec preference 11 g729br8
 codec preference 12 g729r8 bytes 50
```

Related Commands

Command	Description
codec	Specifies the voice coder rate of speech for a dial peer.
voice-class codec	Applies a codec preference list to a specific dial peer.
voice-card	Creates a codec preference list that is independent of a dial peer and can be used on multiple dial peers.

comfort-noise

To generate background noise to fill silent gaps during calls if voice activity detection (VAD) is activated, use the **comfort-noise** command in voice-port configuration mode. To provide silence when the remote party is not speaking and VAD is enabled at the remote end of the connection, use the **no** form of this command.

comfort-noise

no comfort-noise

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **comfort-noise** command to generate background noise to fill silent gaps during calls if VAD is activated. If the **comfort-noise** command is not enabled, and VAD is enabled at the remote end of the connection, the user will hear dead silence when the remote party is not speaking.

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

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

Examples

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

```
voice-port 1/0/0
 comfort-noise
```

Related Commands

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

compand-type

To specify the companding standard used to convert between analog and digital signals in PCM systems on the Cisco MC3810, use the **compand-type** command in voice-port configuration mode. To disable the compand type, use the **no** form of this command.

compand-type {**u-law** | **a-law**}

no compand-type {**u-law** | **a-law**}

Syntax Description

u-law	Specifies the North American u-Law ITU-T PCM encoding standard.
a-law	Specifies the European a-Law ITU-T PCM encoding standard.

Defaults

u-Law (T1 digital)

a-Law (E1 digital)

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was first introduced.

Usage Guidelines

This command applies only to the Cisco MC3810.

Note On the Cisco 3600 series, the u-Law and a-Law settings are configured using the **codec** dial-peer configuration command.

Examples

The following example configures a-law encoding on voice port 1/1 on the Cisco MC3810:

```
voice-port 1/1
 compand-type a-law
```

Related Commands

Command	Description
codec (voice-port configuration)	Configures voice compression on the Cisco MC3810 voice port.

condition

To manipulate the signalling format bit pattern for all voice signalling types on the Cisco MC3810, use the **condition** command in voice-port configuration mode. Use the **no** form of this command to turn off conditioning on the voice-port.

condition {**tx-a-bit** | **tx-b-bit** | **tx-c-bit** | **tx-d-bit**} {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**} {**on** | **off** | **invert**}

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

Syntax Description

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

Defaults

No condition (for all send or receive A, B, C, and D bits)

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to the Cisco MC3810 only.

Use the **condition** command to manipulate the bit patterns sent or received by the Cisco MC3810 to match expected patterns on a connected device. Be careful not to destroy the information content of the bit pattern. For example, forcing the A-bit on or off will prevent FXO interfaces from being able to generate both an on-hook and off-hook state.

Examples

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

```
voice-port 1/1
 condition tx-a-bit invert
 condition rx-a-bit invert
```

Related Commands

Command	Description
define	Defines the send and receive bits for E&M and E&M MEL CAS voice signalling on the Cisco MC3810 multiservice concentrator.
ignore	Specifies the E&M or E&M MEL CAS voice port on the Cisco MC3810 multiservice concentrator to ignore specific receive bits.

connect

To define connections between T1 or E1 controller ports for drop-and-insert (also called TDM cross-connect), use the **connect** command in global configuration mode. Use the **no** form of this command to restore default values.

connect *id* {**t1** | **e1**} *slot/port-1 tdm-group-no-1* {**t1** | **e1**} *slot/port-2 tdm-group-no-2*

no connect *id* {**t1** | **e1**} *slot/port-1 tdm-group-no-1* {**t1** | **e1**} *slot/port-2 tdm-group-no-2*

Syntax Description

id	A name for this connection.
t1	Specifies a T1 port.
e1	Specifies an E1 port.
slot/port-1	The location of the first T1 or E1 controller to be connected. Valid values for *slot* and *port* are 0 and 1.
tdm-group-no-1	The number identifier of the time-division multiplexing (TDM) group associated with the first T1 or E1 controller port and created by using the **tdm-group** command. Valid values are from 0 to 23 for T1 and from 0 to 30 for E1.
slot/port-2	The location of the second T1 or E1 controller port to be connected. Valid values for *slot* are from 0 to 5 depending on the platform. Valid values for *port* are 0 to 3 depending on the platform and the presence of a network module.
tdm-group-no-2	The number identifier of the time-division multiplexing (TDM) group associated with the second T1 or E1 controller and created by using the **tdm-group** command. Valid values are from 0 to 23 for T1 and from 0 to 30 for E1.

Defaults

There is no drop-and-insert connection between the ports.

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	The command was introduced.

Usage Guidelines

The **connect** command creates a named connect between two TDM groups associated with drop-and-insert ports on T1 or E1 interfaces where the user has already defined the groups by using the **tdm-group** command.

Examples

The following example shows how two T1 TDM groups are set up and then connected:

```
Router(config)# controller T1 1/0
Router(config-controller)tdm-group 2 timeslots 13-24 type e&m
Router(config-controller)# controller T1 1/1
Router(config-controller)tdm-group 3 timeslots 13-24 type e&m
Router(config-controller)exit
Router(config)connect tdm1 T1 1/0 2 T1 1/1 3
```

Related Commands

Command	Description
show connect	Displays configuration information about drop-and-insert connections that have been configured on a router.
tdm-group	Configures a list of timeslots for creating clear channel groups (pass-through) for TDM cross-connect.

connect (global)

To configure a data connection on the Cisco MC3810 to an IGX switch that will travel over the FTC trunk, use the **connect** command in global configuration mode. To cancel the connection, use the **no** form of this command.

connect *interface* **dlci** *dlci* *ftc-trunk* **connection-id** *cid*

no connect *interface* **dlci** *dlci* *ftc-trunk* **connection-id** *cid*

Syntax Description

interface	Specifies the interface on the Cisco MC3810 on which the connection is configured.
dlci *dlci*	Specifies the DLCI for the connection.
ftc-trunk	Specifies the Cisco MC3810 interface on which the FTC trunk was configured.
connection-id *cid*	Specifies the connection ID.

Defaults

No data connection over the FTC trunk is configured.

Command Modes

Global configuration

Command History

Release	Modification
12.0(2)T	This command was introduced.

Usage Guidelines

When configuring a connection for an IP routing data connection over an FTC trunk, specify **Switch0** for the local *interface* value.

For this command, when entering the DLCI value, do not use either of the following DLCIs:

- The DLCI that was assigned to the FTC management DLCI using the **ftc-trunk management-dlci** command
- The DLCI that was assigned to the FTC Frame Relay DLCI using the **ftc-trunk frame-relay-dlci** command.

Examples

The following command establishes a data connection on serial port 1:

```
connect Serial1 dlci 250 Serial0:0 connection-id 250
```

Related Commands

Command	Description
connect voice	Configures a connection on the Cisco MC3810 multiservice concentrator between a voice dial peer and the FTC trunk.
encapsulation ftc-trunk	Enables FTC trunk encapsulation on a serial interface on the Cisco MC3810 multiservice concentrator.
ftc-trunk frame-relay-dlci	Maps the FTC Frame Relay DLCI to an existing Frame Relay DLCI for data traffic on the Cisco MC3810.
ftc-trunk management-dlci	Maps the FTC management DLCI to an existing Frame Relay DLCI for management traffic on the Cisco MC3810.
ftc-trunk management-protocol	Selects the mode on the Cisco MC3810 that management frames are sent on the FTC trunk management DLCI.
ftc-trunk voice-dlci	Maps the FTC voice DLCI to an existing Frame Relay DLCI for voice traffic on the Cisco MC3810.

connect voice

To configure a connection on the Cisco MC3810 between a voice dial peer and the FTC trunk, use the **connect voice** command in global configuration mode. To disable the connection, use the **no** form of this command.

connect voice *dial-peer-tag ftc-trunk* **connection-id** *cid*

no connect voice *dial-peer-tag ftc-trunk* **connection-id** *cid*

Syntax Description

dlal-peer-tag	Specifies the voice dial-peer tag for the connection. Valid entries are from 1 to 10000. The dial peer can either be a POTS dial peer or a VoFR dial peer.
ftc-trunk	Specifies the Cisco MC3810 interface on which the FTC trunk was configured.
connection-id *cid*	Specifies the connection ID.

Defaults

No voice connection over the FTC trunk is configured.

Command Modes

Global configuration

Command History

Release	Modification
12.0(2)T	This command was introduced.

Examples

The following command establishes a voice connection for dial peer 2 on serial port 0:

```
connect Voice2 Serial0:0 connection-id 50
```

Related Commands

Command	Description
connect (global)	Configures a data connection on the Cisco MC3810 to a 16x switch that will travel over the FTC trunk.
encapsulation ftc-trunk	Enables FTC trunk encapsulation on a serial interface on the Cisco MC3810 multiservice concentrator.
ftc-trunk frame-relay-dlci	Maps the FTC Frame Relay DLCI to an existing Frame Relay DLCI for data traffic on the Cisco MC3810.
ftc-trunk management-dlci	Maps the FTC management DLCI to an existing Frame Relay DLCI for management traffic on the Cisco MC3810.
ftc-trunk management-protocol	Selects the mode on the Cisco MC3810 that management frames are sent on the FTC trunk management DLCI.
ftc-trunk voice-dlci	Maps the FTC voice DLCI to an existing Frame Relay DLCI for voice traffic on the Cisco MC3810.

connection

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

connection {**plar** | **tie-line** | **plar-opx**} *string* | {**trunk** | *string* [**answer-mode**]}

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

Syntax Description

plar	Specifies a private line auto ring down (PLAR) connection. PLAR is handled by associating a peer directly with an interface; when an interface goes off-hook, the peer is used to set up the second call leg and conference them together without the caller having to dial any digits.
tie-line	(This keyword is specific to the Cisco MC3810.) Specifies a tie-line connection to a private branch exchange (PBX).
plar-opx	(This keyword is specific to the Cisco MC3810) Specifies a PLAR Off-Premises eXtension connection. Using this option, the local voice-port provides a local response before the remote voice-port receives an answer. On FXO interfaces, the voice-port will not answer until the remote side answers.
string	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.
trunk	Specifies a straight tie-line connection to a private branch exchange (PBX).
string	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.
answer-mode	(Optional; used only with the **trunk** keyword.) Specifies that the router should not attempt to initiate a trunk connection, but should wait for an incoming call before establishing the trunk.

Defaults

No connection mode is specified.

Command Modes

Voice-port configuration

Command History

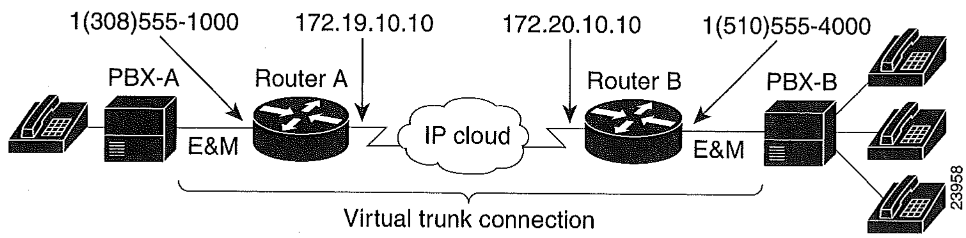
Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA1	This command was first supported on the Cisco MC3810, and the **tie-line** keyword was made available on the Cisco MC3810.
11.3(1)MA5	The **plar-opx** keyword was first made available on the Cisco MC3810 as the **plar-opx-ringrelay**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

Usage Guidelines

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

Use the **connection trunk** command to specify a straight tie-line connection (in other words, a virtual trunk connection) to a PBX. Voice over IP simulates a trunk connection by creating virtual trunk tie lines between PBXs connected to Cisco devices on each side of a VoIP connection. (See Figure 2.) In this example, two PBXs are connected using a virtual trunk. PBX-A is connected to Router A via an E&M voice port; PBX-B is connected to Router B via an E&M voice port. The Cisco routers spoof the connected PBXs into believing that a permanent trunk tie line exists between them.

Figure 2 Virtual Trunk Connection



To configure virtual trunk connections in Voice over IP, the following restrictions apply:

- You can use the following voice port combinations:
  - E&M to E&M (same type)
  - FXS to FXO
  - FXS to FXS (with no signalling)
- Do not perform number expansion on the destination pattern telephone numbers configured for trunk connection.
- Configure both end routers for trunk connections.
- The connected Cisco routers must be Cisco 2600 or Cisco 3600 series routers.

Note Because virtual trunk connections do not support number expansion, the destination patterns on each side of the trunk connection must match exactly.

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

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

VoIP establishes the trunk connection immediately after it is configured. Both ports on either end of the connection are dedicated until you disable trunking for that connection. If for some reason the link between the two switching systems goes down, the virtual trunk re-establishes itself after the link comes back up.

The **connection tie-line** command is used on the Cisco MC3810 when a dial plan requires that additional digits are added in front of any digits dialed by the PBX, and that the combined set of digits are used to route the call via the dial-peers and into the network. The operation is similar to the **connection plar** command operation, but in this case the tie-line port also waits to collect digits from the PBX. The tie-line digits are also automatically stripped by a terminating port.

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

Examples

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

```
voice-port 1/0/0
 connection plar 5559262
```

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

```
voice-port 1/1
 connection tie-line 5559262
```

The following example configures the routers on both sides of a Voice over IP connection (as illustrated in Figure 2) to support trunk connections:

Router A

```
configure terminal
voice-port 1/0/0
 connection trunk +15105554000
dial-peer voice 10 pots
 destination-pattern +13085551001
 port 1/0/0
dial-peer voice 100 voip
 session-target ipv4:172.20.10.10
 destination-pattern +15105554000
```

Router B

```
configure terminal
voice-port 1/0/0
 connection trunk +13085551000
dial-peer voice 20 pots
 destination-pattern +15105554001
 port 1/0/0
dial-peer voice 200 voip
 session-target ipv4:172.19.10.10
 destination-pattern +13085551000
```

Related Commands

Command	Description
session protocol	Establishes a session protocol for calls between the local and remote routers through the packet network in Voice over IP.
voice-port	Enters voice-port configuration mode.

connection-timeout

To configure the time that a connection is maintained after completing a communication exchange, use the **connection-timeout** command in settlement configuration mode. Use the **no** form of this command to reset to the default value of this command.

connection-timeout *num*

no connection-timeout *num*

Syntax Description

num	Time (in seconds) that a connection is maintained after the communication exchange is completed. Values can range from zero (0) to 86400 seconds, zero (0) means forever.

Defaults

The default connection timeout is 3600 seconds (1 hour).

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

When you configure the **connection timeout** command, the router maintains the connection for this period in anticipation of future communication exchanges to the same server.

Examples

The following example configures the connection timeout to be 3600 seconds:

```
settlement 0
   connection timeout 3600
```

Related Commands

Command	Description
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of connection attempts to the provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.

Command	Description
settlement	Enters settlement mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

copy flash vfc

To copy a new version of VCWare from the Cisco AS5300 motherboard to VFC Flash memory, use the **copy flash vfc** command in privileged EXEC mode.

copy flash vfc *slot-number*

Syntax Description

slot-number	Indicates the slot on the Cisco AS5300 where the VFC is installed. Valid entries are from 0 to 2.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use the **copy flash vfc** command to use the standard copy user interface to copy a new version of VCWare from the Cisco AS5300 motherboard to VFC Flash memory. The VFC is a plug-in feature card for the Cisco AS5300 and has its own Flash memory storage for embedded firmware. For more information about VFCs, refer to *Installing Voice over IP Feature Cards in Cisco AS5300 Universal Access Servers*.

Once the VCWare file has been copied, use the **unbundle vfc** command to uncompress and install VCWare.

Examples

The following example copies a new version of VCWare from the Cisco AS5300 motherboard to VFC Flash memory:

```
copy flash vfc 0
```

Related Commands

Command	Description
copy tftp vfc	Copies a new version of VCWare from a TFTP server to VFC Flash memory.
unbundle vfc	Unbundles the current running image of VCWare or DSPWare into separate files.

copy tftp vfc

To copy a new version of VCWare from a TFTP server to VFC Flash memory, use the **copy tftp vfc** command in privileged EXEC mode.

copy tftp vfc *slot-number*

Syntax Description

slot-number	Indicates the slot on the Cisco AS5300 where the VFC is installed. Valid entries are from 0 to 2.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use the **copy tftp vfc** command to copy a new version of VCWare from a TFTP server to VFC Flash memory. The VFC is a plug-in feature card for the Cisco AS5300 and has its own Flash storage for embedded firmware. For more information about VFCs, refer to *Installing Voice Over IP Feature Cards in Cisco AS5300 Universal Access Servers.*

Once the VCWare file has been copied, use the **unbundle vfc** command to uncompress and install VCWare.

Examples

The following example copies a file from the TFTP server to VFC Flash memory:

```
copy tftp vfc 0
```

Related Commands

Command	Description
copy flash vfc	Copies a new version of VCWare from the Cisco AS5300 motherboard to VFC Flash memory.
unbundle vfc	Unbundles the current running image of VCWare or DSPWare into separate files.

cptone

To specify a regional analog voice interface-related tone, ring, and cadence setting, use the **cptone** command in voice-port configuration mode. Use the **no** form of this command to disable the selected tone.

cptone *locale*

no cptone *locale*

ISDN PRI

cptone {**australia** | **brazil** | **china** | **finland** | **france** | **germany** | **japan** | **northamerica** | **sweden** | **unitedkingdom**}

no cptone

E1 R2 signalling

cptone {**australia** | **brazil** | **china** | **finland** | **france** | **germany** | **japan** | **northamerica** | **sweden** | **unitedkingdom**}

no cptone

Syntax Description

locale	Keyword specifying an analog voice interface-related default tone, ring, and cadence setting for a specified country. Valid entries for the Cisco MC3810 prior to release 12.0(4)T are: **argentina**, **australia**, **austria**, **belgium**, **brazil**, **canada**, **china**, **colombia**, **czechrepublic**, **denmark**, **finland**, **france**, **germany**, **greatbritain**, **greece**, **hongkong**, **hungary**, **iceland**, **india**, **indonesia**, **ireland**, **israel**, **italy**, **japan**, **korea**, **luxembourg**, **malaysia**, **mexico**, **netherlands**, **newzealand**, **norway**, **peru**, **philippines**, **poland**, **portugal**, **russia**, **singapore**, **slovakia**, **slovenia**, **southafrica**, **spain**, **sweden**, **switzerland**, **taiwan**, **thailand**, **turkey**, **unitedstates**, and **venezuela**. The Cisco 2600 series, 3600 series and the Cisco MC3810 comply with the ISO 3166 country name standards, which use a two-letter code to represent a country. Valid entries are listed in Table 11.

Table 11 lists valid entries for the *locale* argument.

Table 11 cptone locale Argument Command Entries

cptone Command Entry	Country
ar	Argentina
au	Australia
at	Austria
be	Belgium
br	Brazil

Table 11 cptone locale Argument Command Entries (continued)

cptone Command Entry	Country
ca	Canada
cn	China
co	Colombia
cz	Czech Republic
dk	Denmark
fi	Finland
fr	France
de	Germany
gb	Great Britain
gr	Greece
hk	Hong Kong
hu	Hungary
is	Iceland
in	India
id	Indonesia
ie	Ireland
il	Israel
it	Italy
jp	Japan
kr	Korea Republic
lu	Luxembourg
my	Malaysia
mx	Mexico
nl	Netherlands
nz	New Zealand
no	Norway
pe	Peru
ph	Philippines
pl	Poland
pt	Portugal
ru	Russian Federation
sg	Singapore
sk	Slovakia
si	Slovenia
za	South Africa
es	Spain

Table 11 cptone locale Argument Command Entries (continued)

cptone Command Entry	Country
se	Sweden
ch	Switzerland
tw	Taiwan
th	Thailand
tr	Turkey
gb	Great Britain
us	United States
ve	Venezuela

ISDN PRI.

australia	Specifies an analog voice interface-related default tone, ring, and cadence setting for Australia.
brazil	Specifies an analog voice interface-related default tone, ring, and cadence setting for Brazil.
china	Specifies an analog voice interface-related default tone, ring, and cadence setting for China.
finland	Specifies an analog voice interface-related default tone, ring, and cadence setting for Finland.
france	Specifies an analog voice interface-related default tone, ring, and cadence setting for France.
germany	Specifies an analog voice interface-related default tone, ring, and cadence setting for Germany.
japan	Specifies an analog voice interface-related default tone, ring, and cadence setting for Japan.
northamerica	Specifies an analog voice interface-related default tone, ring, and cadence setting for North America.
sweden	Specifies an analog voice interface-related default tone, ring, and cadence setting for Sweden.
unitedkingdom	Specifies an analog voice interface-related default tone, ring, and cadence setting for the United Kingdom.

E1 R2 signalling:

locale	Keyword specifying an analog voice interface-related default tone, ring, and cadence settings for a specified country. Valid entries are listed in Table 11.

Defaults

northamerica for the Cisco MC3810 for versions prior to Release 12.0(4)T; and for ISDN PRI. **us** for the Cisco MC3810 for 12.0(4)T and higher and for E1 R2 signalling.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA	The full keyword names for the countries were first supported on the Cisco MC3810.
12.0(4)T	Support was added for the ISO 3166 two-letter country codes on the Cisco MC3810.

Usage Guidelines

Use the **cptone** command to specify a regional analog voice interface-related default tone, ring, and cadence setting for a specified voice port. This command only affects the tones generated at the local interface. It does not affect any information passed to the remote end of a connection, or any tones generated at the remote end of a connection.

This command only affects the tones generated at the local interface. It does not affect any information passed to the remote end of a connection, or any tones generated at the remote end of a connection.

If your device is configured to support E1 R2 signalling, the E1 R2 signalling type (whether ITU, ITU variant, or local variant as defined by the **cas-custom** command) needs to match the appropriate PCM encoding type as defined by the **cptone** command. For countries for which a **cptone** value has not yet been defined, you can try the following:

- If the country uses a-Law E1 R2 signalling, use the **gb** value for the **cptone** command.
- If the country uses u-Law E1 R2 signalling, use the **us** value for the **cptone** command.

Examples

The following example configures United States as the call progress tone locale on the Cisco 3600 series, beginning from global configuration mode:

```
voice-port 1/0/0
 cptone us
```

The following example configures Singapore as the call progress tone locale on the Cisco MC3810, beginning from global configuration mode:

```
voice-port 1/1
 cptone sg
```

The following example configures Japan as the call progress tone locale:

```
voice-port 0:D
 cptone japan
```

The following example configures Brazil as the call progress tone locale on the Cisco AS5300:

```
voice-port 1:0
 cptone BR
 description Brasil Tone
```

Related Commands

Command	Description
voice-port	Opens voice-port configuration mode.

cross-connect

To cross-connect two groups of digital signal level 0s (DS0s) from two controllers on the Cisco MC3810, or to cross-connect the Universal I/O (UIO) serial port for pass-through traffic to a trunk controller, use the **cross-connect** command in global configuration command. Use the **no** form of this command to remove the cross-connect function for the given controller.

Pass-through between two controllers:

cross-connect *id controller-1 tdm-group-no-1 controller-2 tdm-group-no-2*

no cross-connect *id controller-1 tdm-group-no-1 controller-2 tdm-group-no-2*

Pass-through traffic from a UIO serial port to a trunk controller:

cross-connect *id interface-serial controller tdm-group-no*

no cross-connect *id interface-serial controller tdm-group-no*

Note The UIO serial port is either serial port 0 or 1.

Syntax Description

For pass-through between two controllers:

id	Unique ID assigned to this cross-connection. The valid range is from 0 to 31.
controller-1	Type of the first controller (T1 0, T1 1, or E1)
tdm-group-no-1	TDM group number assigned to the first controller.
controller-2	Type of the second controller (T1, E1 0, or E1 1).
tdm-group-no-2	TDM group number assigned to the second controller.

For pass-through traffic from a UIO serial port to a trunk controller:

id	Unique ID assigned to this cross connection.
interface-serial	Number of the serial port, either 0 or 1.
controller	Type of the controller. Enter one of the following: T1 0, T1 1, E1 0, or E1 1.
tdm-group-no	TDM group number assigned to the controller.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Examples

The following example configures a pass-through cross-connect from serial port 0 to controller T1 1 on TDM group 20:

```
cross-connect 10 serial0 T1 1 20
```

Related Commands

Command	Description
supervisory disconnect	Configures a list of timeslots for creating clear channel groups (pass-through) for TDM cross-connect.

customer-id

To identify a carrier or internet service provider with a settlement provider, use the **customer-id** command in settlement configuration mode. Use the **no** form of this command to reset to the default value of this command.

customer-id *num*

no customer-id *num*

Syntax Description

num	Customer ID number as provided by the settlement server.

Defaults

The default customer ID is 0.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Examples

The following example identifies a carrier or service provider with the ID number of 1000:

```
settlement 0
    customer id 1000
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of connection attempts to the provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement configuration mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.
url	Configures the ISP address.

default-file vfc

To specify an additional (or different) file from the ones in the default file list and stored in VFC Flash memory, use the **default-file vfc** command in global configuration mode. To delete the file from the default file list, use the **no** form of this command.

default-file *filename* **vfc** *slot*

no default-file *filename* **vfc** *slot*

Syntax Description

filename	Indicates the file to be retrieved from VFC Flash memory and used (as the default file) to boot up the system.
slot	Indicates the slot on the Cisco AS5300 where the VFC is installed. Valid entries are from 0 to 2.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

When VCWare is unbundled, it automatically adds DSPWare to Flash memory, creates both the capability and default file lists, and populates these lists with the default files for that version of VCWare. The default file list includes the files that will be used to boot up the system.

Use the **default-file vfc** command to add a specified file to the default file list, replacing the existing default for that extension type.

Examples

The following example specifies that the bas-vfc-1.0.14.0.bin file, which is stored in VFC Flash memory, be added to the default file list:

```
config term
 default-file bas-vfc-1.0.14.0.bin vfc 0
```

Related Commands

Command	Description
cap-list vfc	Adds a voice codec overlay file to the capability file list.
delete vfc	Deletes a file from VFC Flash memory.

define

To define the transmit and receive bits for E&M and E&M MEL CAS voice signalling on the Cisco MC3810, use the **define** command in voice-port configuration mode. Use the **no** form of this command to restore the default value.

define {**Tx-bits** | **Rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}

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

Syntax Description

Tx-bits	Send signalling bits.
Rx-bits	Receive signalling bits.
seize	Define the pattern that represents the seized state.
idle	Define the pattern that represents the idle state.
0000...1111	Define the appropriate bit pattern.

Defaults

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

For E&M:

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

For E&M MEL CAS:

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

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to digital voice ports on the Cisco MC3810 only.

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

Examples

To configure a voice-port sending traffic in North American E&M signalling format to convert the signalling to Mercury Exchange Limited (MEL) CAS format, enter the following commands:

```
voice-port 1/1
 define rx-bits idle 1101
 define rx-bits idle 0101
 define tx-bits seize 1101
 define tx-bits seize 0101
```

Related Commands

Command	Description
condition	Manipulates the signalling format bit-pattern for all voice signalling types on the Cisco MC3810 multiservice concentrator.
ignore	Specifies the E&M or E&M MEL CAS voice port on the Cisco MC3810 multiservice concentrator to ignore specific receive bits.

delete vfc

To delete a file from VFC Flash memory, use the **delete vfc** command in privileged EXEC mode.

delete *filename* **vfc** *slot*

Syntax Description

filename	Specifies the file in VFC Flash memory to be deleted.
slot	Specifies the slot on the Cisco AS5300 where the specified VFC resides. Valid entries are from 0 to 2.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use the **delete vfc** command to delete a specific file from VFC Flash memory, and to remove the file from the default list or capability list if the specified file is included on those lists.

Note Deleting a file from VFC Flash memory does not free the VFC Flash memory space the file occupied. To free VFC Flash memory space, use the **erase vfc** command.

Examples

The following example deletes the bas-vfc-1.0.14.0.bin file, which is stored in VFC Flash memory of the VFC located in slot 0:

```
delete bas-vfc-1.0.14.0.bin vfc 0
```

Related Commands

Command	Description
default-file vfc	Specifies an additional (or different) file from the ones in the default file list and stored in VFC Flash memory.
show vfc directory	Displays the list of all files residing on this VFC.
erase vfc	Erases the Flash memory of a specified VFC.

description

To include a description of what this voice port is connected to, use the **description** command in voice-port configuration mode. Use the **no** form of this command to disable this feature.

description *string*

no description *string*

Syntax Description

string	Character string from 1 to 255 characters.

Defaults

Enabled with a null string.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to both the Cisco MC3810 and the Cisco 3600 series.

Use the **description** command to include descriptive text about this voice-port connection. This information is displayed when you issue a **show** command and does not affect the operation of the interface in any way.

Examples

The following example identifies voice port 1/0/0 on the Cisco 3600 series as being connected to the Purchasing department:

```
voice-port 1/0/0
 description purchasing_dept
```

The following example identifies voice port 1/1 on the Cisco MC3810 as being connected to the Marketing department:

```
voice-port 1/1
 description marketing_dept
```

description (DSP)

To include a specific description about the digital signal processor (DSP) interface, use the **description** command in DSPfarm interface configuration mode. Use the **no** form of this command to disable this feature.

description *string*

no description *string*

Syntax Description

string	Character string from 1 to 80 characters.

Defaults

Enabled with a null string.

Command Modes

DSPfarm interface configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **description** command to include descriptive text about this DSP interface connection. This information is displayed when you issue a **show** command and does not affect the operation of the interface in any way.

Examples

The following example identifies DSPfarm interface 1/0 on the Cisco 7200 series router as being connected to the marketing department:

```
dspint dspfarm 1/0
 description marketing_dept
```

destination-pattern

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

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

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

Syntax Description

+	(Optional) Character indicating an E.164 standard number. The plus sign (+) is not supported on the Cisco MC3810.
string	Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters: • The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. On the Cisco 3600 only, these characters cannot be used as leading characters in a string (for example, *650). • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit (this character is used as a wildcard). On the Cisco 3600, the period cannot be used as a leading character in a string (for example, .650).
T	(Optional) Control character indicating that the **destination-pattern** value is a variable length dial-string.

Defaults

Enabled with a null string

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.

Usage Guidelines

Use the **destination-pattern** command to define the E.164 telephone number for a dial peer.

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

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

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

Examples

The following example configures the E.164 telephone number, 555-7922, for a dial peer:

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

Related Commands

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

destination-pattern (ISDN)

To specify the ISDN directory number for the telephone interface, use the **destination-pattern** command in interface configuration mode. Use the **no** form of this command to disable the specified ISDN directory number.

destination-pattern *isdn*

no destination-pattern

Syntax Description

isdn	Local ISDN directory number assigned by your telephone service provider.

Defaults

A default ISDN directory number is not defined for this interface.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command is applicable to the Cisco 800 series routers.

You must specify this command when creating a dial peer. This command will not work if it is not specified within the context of a dial peer. For information on creating a dial peer, refer to the *Cisco 800 Series Routers Software Configuration Guide*.

Do not specify an area code with the local ISDN directory number.

Examples

The following example specifies 555-1111 as the local ISDN directory number:

```
destination-pattern 5551111
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
no call-waiting	Disables call waiting.
port (dial-peer)	Enables an interface on a PA-4R-DTR port adapter to operate as a concentrator port.
ring	Sets up a distinctive ring for telephones, fax machines, or modems connected to a Cisco 800 series router.
show dial-peer voice	Displays configuration information and call statistics for dial peers.

device-id

To specify a gateway associated with a settlement provider, use the **device-id** command in settlement configuration mode. Use the **no** form of this command to reset to the default value of this command.

device-id *num*

no device-id *num*

Syntax Description

num	Device ID number as provided by the settlement server.

Defaults

The default device ID is 0.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Examples

The following example specifies gateway with device ID# 1000 associated with the settlement provider:

```
settlement 0
   device-id 1000
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of connection attempts to the provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement configuration mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.
url	Configures the ISP address.

dial-control-mib

To specify attributes for the call history table, use the **dial-control-mib** command in global configuration mode.

dial-control-mib {**max-size** *number* | **retain-timer** *number*}

Syntax Description

max-size *number*	Specifies the maximum size of the call history table. Valid entries are from 0 to 500 table entries. A value of 0 prevents any history from being retained.
retain-timer *number*	Specifies the length of time, in minutes, for entries in the call history table. Valid entries are from 0 to 2147483647 minutes. A value of 0 prevents any history from being retained.

Defaults

The default call history table length is 50 table entries. The default retain timer is 15 minutes.

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 Series routers.
12.0(1)XA and 12.0(2)T	This command was first applied to the CDR feature on the Cisco MC3810.

Examples

The following example configures the call history table to hold 400 entries, with each entry remaining in the table for 10 minutes:

```
configure terminal
 dial-control-mib max-size 400
 dial-control-mib retain-timer 10
```

dial-peer terminator

To designate a special character to be used as a terminator for variable length dialed numbers, use the **dial-peer terminator** command in global configuration mode. Use the **no** form of this command to disable the designated terminating character.

dial-peer terminator *character*

no dial-peer terminator *character*

Syntax Description

character	Designates the terminating character for a variable-length dialed number. Valid numbers and characters are #, *, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, a, b, c, and d.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0	This command was introduced.

Usage Guidelines

There are certain areas in the world (for example, in certain European countries) where valid telephone numbers can vary in length. When a dialed-number string has been identified as a variable length dialed-number, the system waits until the configured value for the **timeouts interdigits** command has expired before placing the call. To avoid waiting until the interdigit timeout value has expired, you can designate a special character as a terminator—meaning that when you dial that character, the system no longer waits for any additional digits and places the call. Use the **dial-peer terminator** global configuration command to designate a particular character as a terminator.

Examples

The following example configures # as the special terminating character for variable-length dialed-numbers:

```
configure terminal
 dial-peer terminator #
```

Related Commands

Command	Description
answer-address	Specifies the full E.164 telephone number to be used to identify the dial peer of an incoming call.
destination-pattern	Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.

dial-peer video

To define a video ATM dial peer for a local or remote video codec, specify video-related encapsulation, and enter dial-peer configuration mode, enter the **dial-peer video** command in global configuration mode. The video dial peer is persistent and remains until you use the **no** form of the command to remove it.

dial-peer video *tag* {**videocodec** | **videoatm**}

no dial-peer video *tag* {**videocodec** | **videoatm**}

Syntax Description

tag	Digits defining a particular dial peer. Defines the dial peer and assigns the protocol type to the peer. Valid entries are from 1 to 10000. The tag must be unique on the router.
videocodec	This keyword specifies a local video codec connected to the router.
videoatm	This keyword specifies a remote video codec on the ATM network.

Defaults

No video dial peer is configured.

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for ATM interface configuration on the Cisco MC3810.

Usage Guidelines

The *tag* value that you assign must be unique to the device.

Video dial peers are persistent and remain until explicitly removed using the **no** form of the command.

Examples

On a Cisco MC3810, the following example shows the setup of a local video dial peer designated as 10:

```
dial-peer video 10 videocodec
```

Related Commands

Command	Description
show dial-peer video	Displays dial-peer configuration.

dial-peer voice

To enter dial-peer configuration mode (and specify the method of voice-related encapsulation), use the **dial-peer voice** command in global configuration mode. Use the **no** form of this command to disable a defined dial peer.

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

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

Cisco AS5300 access servers

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

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

Cisco 7200 series routers

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

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

Cisco MC3810

dial-peer voice *number* {**pots** | **voatm** | **vofr** | **vohdlc**}

no dial-peer voice *number* {**pots** | **voatm** | **vofr** | **vohdlc**}

Syntax Description

number	Digit(s) that define a particular dial peer. Valid entries are 1 to 2147483647.
mmoip	Indicates that this is a Multimedia Mail peer using IP encapsulation on the IP backbone.
pots	Indicates that this is a POTS peer using Voice over IP encapsulation on the IP backbone.
voatm	Specifies that this is a Voice over ATM dial peer using the real-time AAL5 voice encapsulation on the ATM backbone network.
vofr	Specifies that this is a Voice over Frame Relay dial peer using FRF.11 encapsulation on the Frame Relay backbone network.
vohdlc	Specifies that this is a Voice over HDLC dial peer using Cisco serial encapsulation (HDLC) for voice.
voip	Indicates that this is a VoIP peer using voice encapsulation on the POTS network.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA	This command was first supported on the Cisco MC3810, with support for the **pots**, **vofr**, **voatm**, and **vohdlc** keywords.
12.0(3)T	This command was first supported on the AS5300, with support for the **pots** and **voip** keywords.
12.0(3)XG and 12.0(4)(T)	The **vofr** keyword was added for support for the Cisco 2600 series and Cisco 3600 series platforms.
12.0(4)T	Added **vofr** keyword support for the Cisco 7200 series platform.
12.0(4)XJ	Added **mmoip** keyword support for the Cisco AS5300 platform.

Usage Guidelines

Use the **dial-peer voice** global configuration command to switch to dial-peer configuration mode from global configuration mode and to define a particular dial peer. Use the **exit** command to exit dial-peer configuration mode and return to global configuration mode.

After you have created a dial peer, that dial peer remains defined and active until you disable that particular dial peer. To disable a dial peer, use the **no** form of this command.

In Store and Forward Fax on the Cisco AS5300, the POTS dial peer defines the inbound faxing line characteristics from the sending fax device to the receiving Cisco AS5300 and the outbound line characteristics from the sending Cisco AS5300 to the receiving fax device. The MMoIP dial peer defines the inbound faxing line characteristics from the Cisco AS5300 to the receiving SMTP mail server. This command applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

The following example shows how to access dial-peer configuration mode and configure a POTS peer identified as dial peer 100, starting from global configuration mode:

```
configure terminal
 dial-peer voice 100 pots
```

The following example accesses dial-peer configuration mode and configures a POTS peer identified as dial peer 10 and an MMoIP dial peer identified as dial peer 20:

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

The following example disables the MMoIP peer identified as dial peer 20:

```
configure terminal
 no dial-peer voice 20
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.

Command	Description
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
sequence-numbers	Enables the generation of sequence numbers in each frame generated by the DSP for Voice over Frame Relay applications.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
voice-port	Opens voice-port configuration mode.

dial-type

To specify the type of out-dialing for voice port interfaces, use the **dial-type** command in voice-port configuration mode. Use the **no** form of this command to disable the selected type of dialing.

dial-type {**dtmf** | **pulse**}

no dial-type {**dtmf** | **pulse**}

Syntax Description

dtmf	Specifies a touch-tone dialer.
pulse	Specifies a pulse dialer.

Defaults

dtmf

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to both the Cisco MC3810 and the Cisco 3600 series.

Use the **dial-type** command to specify an out-dialing type for an FXO or E&M voice port interface; this command is not applicable to FXS voice ports because they do not generate out-dialing. Voice ports can always detect dtmf and pulse signals. This command does not affect voice port dialing detection.

The **dial-type** command affects out-dialing as configured for the dial peer.

Examples

The following example configures a voice port on the Cisco 3600 series to support a touch-tone dialer:

```
voice-port 1/0/0
 dial-type dtmf
```

The following example configures a voice port on the Cisco MC3810 to support a rotary (pulse tone) dialer:

```
voice-port 1/1
 dial-type pulse
```

direct-inward-dial

To enable the direct inward dial (DID) call treatment for the incoming called number, use the **direct-inward-dial** command in dial-peer configuration mode. Use the **no** form of this command to disable DID.

direct-inward-dial

no direct-inward-dial

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)NA	This command was introduced.
12.0(4)T	This command was modified for Store and Forward Fax.

Usage Guidelines

Use the **direct-inward-dial** command to enable the DID call treatment for the incoming called numbers. When this feature is enabled, the incoming call is treated as if the digits are received from the DID trunk. The called number is used to select the outgoing dial peer. No dial tone will be presented to the caller.

Use the **no** form of this command to disable DID. When disabled, the called number is used to select the outgoing dial peer. The caller will be prompted for a called number via dial tone.

This command is only applicable to POTS dial peers. This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example enables DID call treatment for incoming called numbers:

```
dial peer voice 10 pots
 direct-inward-dial
```

dsn

To specify that a delivery status notice be delivered to the sender, use the **dsn** command in dial-peer configuration mode. Use the **no** form of this command to cancel a specific delay status notice option.

dsn {**delay** | **failure** | **success**}

no dsn {**delay** | **failure** | **success**}

Syntax Description

delay	Indicates that when the mail is sent, the next-hop mailer is requested to send an indication to the FROM address if the mail message is delayed. The definition of delay is made by each mailer and is not controllable by the sender (the AS5300). Each mailer in the path to the recipient that supports the DSN extension receives the same request.
failure	Indicates that when the mail is sent, the next-hop mailer is requested to send a message to the FROM address if the mail message failed to be delivered. Each mailer in the path to the recipient that supports the DSN extension receives the same request.
success	Indicates that when the mail is sent, the next-hop mailer is requested to send a message to the FROM address if the mail message is successfully delivered to the recipient. Each mailer in the path to the recipient that supports the DSN extension receives the same request.

Note In the absence of any other DSN settings ("no dsn," or a mailer in the path that does not support the DSN extension) a failure to deliver will always cause a nondelivery message to be generated. This nondelivery message is colloquially termed a "bounce."

Defaults

The default is **success** and **failure**.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

This command is applicable to MMoIP dial peers.

Delivery status notifications (DSNs) are messages or responses that are automatically generated and sent to the sender or originator of an e-mail message by the SMTP server, notifying the sender of the status of the e-mail message. Specifications for DSN are described in RFC 1891, RFC 1892, RFC 1893, and RFC 1894.

The on-ramp DSN request is included as part of the fax-mail message sent by the on-ramp gateway when the matching MMoIP dial peer has been configured. The on-ramp DSN response is generated by the SMTP server when the fax-mail message is accepted. The DSN is sent back to the user defined in using the **mta send mail-from** command. The off-ramp DSN is requested by the e-mail client. The DSN response is generated by the SMTP server when it receives a request as part of the fax-mail message.

Note DSNs can only be generated if the mail client on the SMTP server is capable of responding to a DSN request.

Because the SMTP server generates the DSNs, you need to configure both the **mail from:** and **rcpt to:** commands for the DSN feature to be operational, for example:

```
mail from: <user@mail-server.company.com>
rcpt to: <fax=555-1212@company.com> NOTIFY=SUCCESS,FAILURE,DELAY
```

There are three different states that can be reported back to the sender:

- Delay—Indicates that, for some reason, the message was delayed in being delivered to the recipient.
- Success—Indicates that the message was successfully delivered to the recipient's mailbox.
- Failure—Indicates that, for some reason, the SMTP server was unable to deliver the message to the recipient.

Because these delivery states are not mutually exclusive, you can configure Store and Forward Fax to generate these messages for all or any combination of these events.

DSN messages notify the sender of the status of a particular e-mail message containing a fax TIFF image. Use the **dsn** command to specify which notification messages will be sent to the user.

The **dsn** command allows you to select more than one notification option by reissuing the command, specifying a different notification option each time. To discontinue a specific notification option, use the **no** form of the command for that specific keyword.

Note If the keyword **failure** is not included when configuring DSN, the sender will receive absolutely no notification of message delivery failure. As a failure is usually significant, care should be taken to always include **failure** as part of the **dsn** command configuration.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example specifies that a DSN message be returned to the sender when the e-mail message containing the fax has been successfully delivered to the recipient or if the message containing the fax has failed, for whatever reason, to be delivered:

```
dial-peer voice 10 mmoip
 dsn success
 dsn failure
```

The following example specifies that a DSN message be returned to the sender either when the e-mail message containing the fax has been successfully delivered to the recipient or when the message has been delayed:

```
dial-peer voice 10 mmoip
 dsn success
 dsn delayed
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from or the Return-Path address).

ds0-group

To define T1/E1 channels for compressed voice calls and the channel-associated signalling (CAS) method by which the router connects to the PBX or PSTN, use the **ds0-group** command in controller configuration mode. The **no** form of the command removes the group and signalling setting.

ds0-group *ds0-group-no* **timeslots** *timeslot-list* **type** {**e&m-immediate** | **e&m-delay** | **e&m-wink** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

no ds0-group *ds0-group-no* **timeslots** *timeslot-list* **type** {**e&m-immediate** | **e&m-delay** | **e&m-wink** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

Syntax Description

ds0-group-no	A value from 0 to 23 that identifies the DS0 group.
timeslots *timeslot-list*	**Timeslot** *timeslot-list* is a single time slot number, a single range of numbers, or multiple ranges of numbers separated by commas. For T1/E1, allowable values are from 1 to 24. Examples are: • 2 • 1-15, 17-24 • 1-23 • 2, 4, 6-12
type	The signalling method selection for the **type** keyword depends on the connection that you are making. The E&M interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The FXS interface allows connection of basic telephone equipment and PBX. The FXO interface is for connecting the central office (CO) to a standard PBX interface where permitted by local regulations; it is often used for OPXs. The options are as follows: • **e&m-immediate** specifies no specific off-hook and on-hook signalling. • **e&m-delay** specifies that the originating endpoint sends an off-hook signal and then and waits for an off-hook signal followed by an on-hook signal from the destination. • **e&m-wink** specifies that the originating endpoint sends an off-hook signal and waits for a wink signal from the destination. • **fxs-ground-start** specifies Foreign Exchange Station ground-start signalling support. • **fxs-loop-start** specifies Foreign Exchange Station loop-start signalling support. • **fxo-ground-start** specifies Foreign Exchange Office ground-start signalling support. • **fxo-loop-start** specifies Foreign Exchange Office loop-start signalling support.

Defaults

No DS0 group is defined.

Command Modes

Controller configuration

Command History

Release	Modification
11.2	This command was introduced for the Cisco AS5300 as the **cas-group** command.
12.0(1)T	The **cas-group** command was introduced for the Cisco 3600 series.
12.0(5)XE and 12.0(7)T	The command was renamed ds0-group on the Cisco AS5300 and on the Cisco 2600 and 3600 series (Digital T1 Packet Voice Trunk Network Modules are required).

Usage Guidelines

The **ds0-group** command automatically creates a logical voice port that is numbered as follows on Cisco 2600 and 3600 series routers: *slot/port:ds0-group-no*. Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

Examples

The following example configures ranges of T1 controller time slots for FXS ground-start and FXO loop-start signalling:

```
controller T1 1/0
 cablelength long 0db
 ds0-group 1 timeslots 4-5 type e&m-immediate-start
```

Related Commands

Command	Description
codec	Specifies the voice coder rate of speech for a dial peer.
codec complexity	Specifies call density and codec complexity based on the codec standard you are using.

dspfarm

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

dspfarm *slot/port*

Syntax Description

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

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XE	This command was introduced.

Examples

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

```
dspint dspfarm 1/0
```

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** command in dial-peer configuration mode. Use the **no** form of this command to remove all signalling options and send the DTMF tones as part of the audio stream.

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

no dtmf-relay [**cisco-rtp**] [**h245-alphanumeric**] [**h245-signal**]

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(1)T	This command was introduced.
12.0(2)XH	This command was modified to include the **h245-signal** keyword.
12.0(5)T	This command was modified for H.323 V2.

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

1. cisco-rtp (highest priority)
2. h245-signal
3. h245-alphanumeric
4. None—DTMF sent in-band

The principal advantage of the **dtmf-relay** command is that it sends 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:

```
configure terminal
 dial-peer voice 103 voip
  dtmf-relay cisco-rtp
  end
```

The next example configures the **dtmf-relay** command for cisco-rtp or h245-signal when sending to dial-peer 103:

```
configure terminal
 dial-peer voice 103 voip
  dtmf-relay cisco-rtp h245-signal
  end
```

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

```
configure terminal
 dial-peer voice 103 voip
  no dtmf-relay
  end
```

Related Commands

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

dtmf-relay (Voice over Frame Relay)

To enable the generation of FRF.11 Annex A frames for a dial peer, use the **dtmf-relay** command in dial-peer configuration mode. Use the **no** form of this command to disable the generation of FRF.11 Annex A frames and return to the default handling of dial digits.

dtmf-relay

no dtmf-relay

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

This command applies to all Voice over Frame Relay, Voice over ATM, and Voice over HDLC applications on the Cisco MC3810, and to Voice over Frame Relay applications on the Cisco 2600 series and 3600 series routers.

Cisco recommends that this command be used with low bit-rate codecs.

When **dtmf-relay** (Voice over Frame Relay) is enabled, the digital signal processor (DSP) generates Annex A frames instead of passing a DTMF tone through the network as a voice sample. For information about the payload format of FRF.11 Annex A frames, see Annex A—Dialed Digit Transfer Syntax, in *Voice over Frame Relay Implementation Agreement—FRF.11*.

Examples

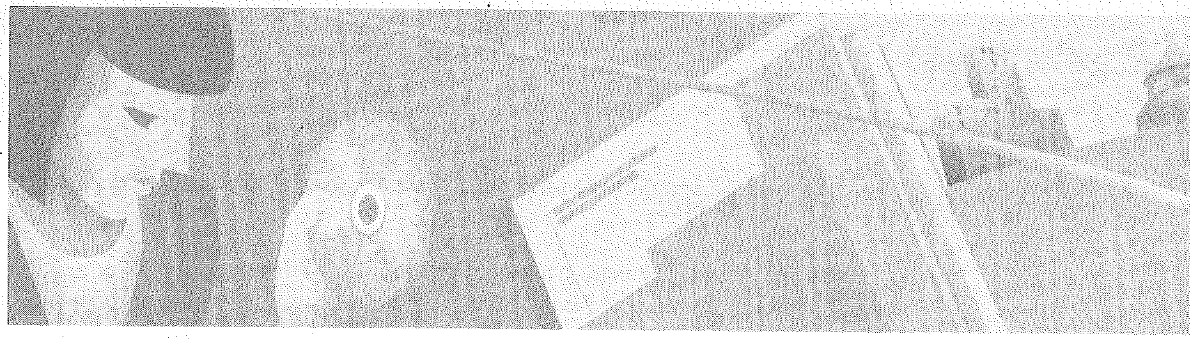
The following example shows how to enable FRF.11 Annex A frames on a Cisco 2600 series or 3600 series router or on an MC3810 concentrator for VoFR dial peer 200, starting from global configuration mode:

```
dial-peer voice 200 vofr
 dtmf-relay
```

Related Commands

Command	Description
called-number (dial-peer)	Enables an incoming VoFR call leg to get bridged to the correct POTS call leg when using a static FRF.11 trunk connection.
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
connection	Specifies a connection mode for a voice port.

Command	Description
cptone	Specifies a regional analog voice interface-related tone, ring, and cadence setting.
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
signal-type	Sets the signalling type to be used when connecting to a dial peer.



Multiservice Applications Commands: E through P

This book documents commands used to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features. Commands in this book are listed alphabetically. For information on how to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features, refer to the *Cisco IOS Multiservice Applications Configuration Guide*.

echo-cancel coverage

To adjust the size of the echo canceller, use the **echo-cancel coverage** command in voice-port configuration mode. Use the **no** form of this command to reset this command to the default value.

echo-cancel coverage {**8** | **16** | **24** | **32**}

no echo-cancel coverage {**8** | **16** | **24** | **32**}

Syntax Description

8	8 milliseconds
16	16 milliseconds
24	24 milliseconds
32	24 milliseconds

Defaults

16 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	The command was introduced.
12.0(5)XK and 12.0(7)T	The command was modified to add the 8-millisecond option.

Usage Guidelines

Use the **echo-cancel coverage** command to adjust the coverage size of the echo canceller. This command enables cancellation of voice that is sent out the interface and received back on the same interface within the configured amount of time. If the local loop (the distance from the interface to the connected equipment producing the echo) is longer, the configured value of this command should be extended.

If you configure a longer value for this command, it takes the echo canceller longer to converge; in this case, the user might hear slight echo when the connection is initially set up. If the configured value for this command is too short, the user might hear some echo for the duration of the call because the echo canceller is not cancelling the longer delay echoes.

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

Note This command is valid only if the echo cancel feature has been enabled. For more information, see the **echo-cancel enable** command.

Examples

The following example adjusts the size of the echo canceller to 8 milliseconds on the Cisco 3600 series:

```
voice-port 1/0:0
 echo-cancel enable
 echo-cancel coverage 8
```

Related Commands

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

echo-cancel enable

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

echo-cancel enable

no echo-cancel enable

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled for all interface types.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to both the Cisco 3600 series and the Cisco MC3810.

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

The **echo-cancel enable** command does not affect the echo heard by the user on the analog side of the connection.

There is no echo path for a 4-wire E&M interface. The echo canceller should be disabled for that interface type.

Note This command is valid only if the **echo-cancel coverage** command has been configured. For more information, refer to the **echo-cancel coverage** command.

Examples

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

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

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

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

Related Commands

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

encapsulation atm-ces

To enable circuit emulation service (CES) ATM encapsulation on the Cisco MC3810, use the **encapsulation atm-ces** command in interface configuration mode. Use the **no** form of this command to disable CES ATM encapsulation.

encapsulation atm-ces

no encapsulation atm-ces

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to ATM configuration on the Cisco MC3810.

This command is only supported on serial ports 0 and 1.

Examples

The following example enables CES ATM encapsulation on serial port 0 on the Cisco MC3810:

```
interface serial 0
 encapsulation atm-ces
```

Related Commands

Command	Description
ces cell-loss-integration-period	Sets the CES cell-loss integration period on the Cisco MC3810 multiservice concentrator.
ces clockmode synchronous	Configures the ATM CES synchronous clock mode on the Cisco MC3810 multiservice concentrator.
ces connect	Maps the CES service to an ATM PVC on the Cisco MC3810 multiservice concentrator.
ces initial-delay	Configures the size of the receive buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.
ces max-buf-size	Configures the send buffer of a CES circuit on the Cisco MC3810 multiservice concentrator.

Command	Description
ces partial-fill	Configures the number of user octets per cell for the ATM CES on the Cisco MC3810 multiservice concentrator.
ces service	Configures the ATM CES type on the Cisco MC3810 multiservice concentrator.

encapsulation ftc-trunk

To enable FTC trunk encapsulation on a serial interface on the Cisco MC3810, use the **encapsulation ftc-trunk** command in interface configuration mode. To disable FTC trunk encapsulation, use the **no** form of this command.

encapsulation ftc-trunk

no encapsulation ftc-trunk

Syntax Description

This command has no keywords or arguments.

Defaults

FTC trunk encapsulation is not enabled.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(2)T	This command was introduced.

Examples

To configure FTC trunk encapsulation on serial interface 1 on the Cisco MC3810, enter the following commands:

```
interface serial 1
 encapsulation ftc-trunk
```

Related Commands

Command	Description
connect (global)	Configures a data connection on the Cisco MC3810 to a 16x switch that will travel over the FTC trunk.
connect voice	Configures a connection on the Cisco MC3810 multiservice concentrator between a voice dial peer and the FTC trunk.
ftc-trunk frame-relay-dlci	Maps the FTC Frame Relay DLCI to an existing Frame Relay DLCI for data traffic on the Cisco MC3810.
ftc-trunk management-dlci	Maps the FTC management DLCI to an existing Frame Relay DLCI for management traffic on the Cisco MC3810.
ftc-trunk management-protocol	Selects the mode on the Cisco MC3810 that management frames are sent on the FTC trunk management DLCI.
ftc-trunk voice-dlci	Maps the FTC voice DLCI to an existing Frame Relay DLCI for voice traffic on the Cisco MC3810.

encryption

To set the encryption method to be negotiated with the provider, use the **encryption** command in settlement configuration mode. Use the **no** form of this command to reset to the default value of this command.

encryption {**des-cbc-sha** | **des40-cbc-sha** | **dh-des-cbc-sha** | **dh-des40-cbc-sha** | **null-md5** | **null-sha**}

no encryption {**des-cbc-sha** | **des40-cbc-sha** | **dh-des-cbc-sha** | **dh-des40-cbc-sha** | **null-md5** | **null-sha**}

Syntax Description

des-cbc-sha	Encryption type SSL_RSA_with_DES_CBC_SHA cipher suite.
des40-cbc-sha	Encryption type SSL_RSA_EXPORT_with_DES40_CBC_SHA cipher suite.
dh-des-cbc-sha	Encryption type SSL_DH_RSA_with_DES_CBC_SHA cipher suite.
dh-des40-cbc-sha	Encryption type SSL_DH_RSA_EXPORT_with_DES40_CBC_SHA cipher suite.
null-md5	Encryption type SSL_RSA_with_NULL_MD5 cipher suite.
null-sha	Encryption type SSL_RSA_with_NULL_SHA cipher suite.

Defaults

The default encryption method is **all**.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

If none of the encryption methods are configured, then the system configures to use all of the encryption methods in the SSL session negotiation.

Examples

The following example configures the encryption type SSL_RSA_with_DES_CBC_SHA cipher suite:

```
settlement 0
 encryption des-cbc-sha
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.

Command	Description
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of connection attempts to the provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement configuration mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.
url	Configures the ISP address.

erase vfc

To erase the Flash memory of a specified voice feature card (VFC), use the **erase vfc** command in privileged EXEC mode.

erase vfc *slot*

Syntax Description

slot	Specifies the slot on the Cisco AS5300 where the specified VFC resides. Valid entries are from 0 to 2.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use the **erase vfc** command to erase the contents of the Flash memory (thereby freeing space in VFC Flash memory) for a specified VFC, including the default file list and the capability file list.

Examples

The following example erases the Flash memory on the VFC located in slot 0:

```
erase vfc 0
```

Related Commands

Command	Description
delete vfc	Deletes a file from VFC Flash memory.

expect-factor

To specify when the router will generate an alarm to the network manager, indicating that the expected quality of voice has dropped, use the **expect-factor** command in dial-peer configuration mode. Use the **no** form of this command to reset the default value.

expect-factor *value*

no expect-factor *value*

Syntax Description

value	Integers that represent the ITU specification for quality of voice as described in G.113. Valid entries are from 0 to 20, with 0 representing toll quality.

Defaults

10

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 3600 series VoIP peers.

Voice over IP monitors the quality of voice received over the network. Use the **expect-factor** command to specify when the router will generate an SNMP trap to the network manager.

Examples

The following example configures toll quality of voice when connecting to a dial peer:

```
dial-peer voice 10 voip
 expect-factor 0
```

fax-rate

To establish the rate at which a fax will be sent to the specified dial peer, use the **fax-rate** command in dial-peer configuration mode. Use the **no** form of this command to reset the dial peer for voice calls.

fax-rate {**1200** | **2400** | **4800** | **7200** | **9600** | **14400** | **disable** | **voice**} *bytes*

no fax-rate

Syntax Description

1200	Specifies a fax transmission speed of 1200 bits per second (bps).
2400	Specifies a fax transmission speed of 2400 bps.
4800	Specifies a fax transmission speed of 4800 bps.
7200	Specifies a fax transmission speed of 7200 bps.
9600	Specifies a fax transmission speed of 9600 bps.
14400	Specifies a fax transmission speed of 14,400 bps.
disable	Disables fax relay transmission capability.
voice	Specifies the highest possible transmission speed allowed by the voice rate.
bytes	(Optional) Forwards DTMF tones by using the H.245 "signal" User Input Indication method. Valid field entries are 0-9, *, #, and A-D.

Defaults

voice

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(2)XH	The fax transmission rate of 12000 was added.
12.0(4)T	This command was first supported on the Cisco MC3810.

Usage Guidelines

Use this command to specify the fax transmission rate to the specified dial peer.

The values for this command apply only to the fax transmission speed and do not affect the quality of the fax itself. The higher values provide a faster transmission speed but monopolize a significantly larger portion of the available bandwidth. Slower transmission speeds use less bandwidth.

If the fax-rate transmission speed is set higher than the codec rate in the same dial peer, the data sent over the network for fax transmission will be above the bandwidth reserved for RSVP. Because more network bandwidth will be monopolized by the fax transmission, Cisco does not recommend setting the fax-rate value higher than the value of the selected codec. If the fax-rate value is set lower than the codec value, faxes will take longer to send but will use less bandwidth.

Examples

The following example configures a transmission speed of 9600 bps for faxes sent to a dial peer:

```
dial-peer voice 100 voip
 fax-rate 9600
```

The following example sets the fax rate at 12000 bits per second and the size of the fax-data frame to 200 bytes:

```
fax-rate 12000 bytes 200
```

Related Commands

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

fax receive called-subscriber

To define the called subscriber identifier (CSI), use the **fax receive called-subscriber** command in global configuration mode. Use the **no** form of this command to disable the configured number.

fax receive called-subscriber {**d**| *string*}

no fax receive called-subscriber {**d**| *string*}

Syntax Description

d	Wildcard specifies that the information displayed is captured from the configured destination pattern.
string	Specifies the destination telephone number. Valid entries are the plus sign (+), numbers 0 through 9, and the space character. This string can specify an E.164 telephone number; if you choose to configure an E.164 telephone number, you must use the plus sign as the first character.

Defaults

Enabled with a null string.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Use the **fax called-subscriber** command to define the number displayed in the LCD of the sending fax device when you are sending a fax to a recipient. Typically, with a standard Group 3 fax device, this is the telephone number associated with the receiving fax device. The command defines the CSI.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example configures 555-1234 as the called-subscriber number:

```
configure terminal
 fax received called-subscriber 5551234
```

fax send center-header

To specify the data that will appear in the center position of the fax header information, use the **fax send center-header** command in global configuration mode. Use the **no** form of this command to disable the selected options.

fax send center-header {**a**| **d**| **p**| **s**| **t**| *string*}

no fax send center-header {**a**| **d**| **p**| **s**| **t**| *string*}

Syntax Description

a	Wildcard that inserts the date in the selected position.
d	Wildcard that inserts the destination address in the selected position.
p	Wildcard that inserts the page count in the selected position.
s	Wildcard that inserts the sender address in the selected position.
t	Wildcard that inserts the transmission time in the selected position.
string	Text string that provides personalized information. Valid characters are any text plus wildcards—for example, Time:t.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Mail messages that contain only text or contain text attachments (MIME media type of text) can be converted by the off-ramp AS5300 into a format understood by fax machines using the Cisco AS5300's text-to-fax converter. When this conversion is performed, this **fax send center-header** command is used to indicate what header information should be added to the center top those pages.

Mail messages with TIFF attachments (MIME media type of image and subtype of TIFF) are expected to include their own per-page headers, and the Cisco AS5300 software does not modify TIFF attachments.

Note Because the Cisco AS5300 does not alter fax TIFF attachments, you cannot configure faxed header information for faxes being converted from TIFF files to standard fax transmissions.

This command lets you configure multiple options at once—meaning that you can combine one or more wild cards with text string information to personalize your fax header information.

Note If the information you have selected for the **fax send center-header** command exceeds the space allocated for the center fax header, the information is truncated.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example selects the transmission time of the fax as the center fax header information:

```
configure terminal
 fax send center-header $t$
```

The following example configures the company name Widget and its address as the center fax header information:

```
configure terminal
 fax send center-header widget $s$
```

Related Commands

Command	Description
fax send left-header	Specifies the data that will appear on the left in the fax header.
fax send right-header	Specifies the data that will appear on the right in the fax header information.

fax send coverpage comment

To define personalized text for the title field of a fax cover sheet, use the **fax send coverpage comment** command in global configuration mode. Use the **no** form of this command to disable the defined comment.

fax send coverpage comment *string*

no fax send coverpage comment *string*

Syntax Description

string	Text string that adds personalized text in the title field of the fax cover sheet. Valid characters are any ASCII characters.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

The **fax send coverpage comment** command can be overridden by the **fax send coverpage e-mail controllable** command.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example configures an individualized title comment of Acme Fax Services for generated fax cover sheets:

```
configure terminal
 fax send coverpage enable
 fax send coverpage comment Acme Fax Services
```

Related Commands

Command	Description
fax send coverpage e-mail-controllable	Defers to the cover page setting in the e-mail header to generate a standard fax cover sheet.
fax send coverpage enable	Enables the Cisco AS5300 to generate fax cover sheets for faxes that originate from e-mail messages.
fax send coverpage show-detail	Prints all of the e-mail header information as part of the fax cover sheet.

fax send coverpage e-mail-controllable

To defer to the cover page setting in the e-mail header to generate a standard fax cover sheet, use the **fax send coverpage e-mail-controllable** command in global configuration mode. Use the **no** form of this command to disable standard fax sheet generation.

fax send coverpage e-mail-controllable

no fax send coverpage e-mail-controllable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

You can also use the destination address of an e-mail message to control the cover page generation on a per-recipient basis. Use the **fax send coverpage e-mail-controllable** command to configure the router to defer to the cover page setting in the e-mail header.

In essence, the off-ramp router defers to the setting configured in the e-mail address itself. For example, if the address has a parameter set to cover=no, this parameter will override the setting for the **fax send coverpage enable** command and the off-ramp gateway will not generate and send a fax cover page. If the address has a parameter set to cover=yes, the off-ramp gateway will defer to this and generate and send a fax cover page.

This command applies to off-ramp Store and Forward Fax functions.

Table 12 shows examples of what the user would enter in the To: field of the e-mail message.

Table 12 Example Entries for the To: Field

Example for To: Field Entries	Description
`FAX=+1-312-555-3260@fax.com`	Fax sent to an E.164-compliant long distance telephone number in the United States. If the **fax coverpage enable** command has been configured, Store and Forward Fax will generate a fax cover page.

Table 12 Example Entries for the To: Field (continued)

Example for To: Field Entries	Description
`FAX=+1-312-555-3260/cover=no@fax.com`	Fax sent to an E.164-compliant long distance telephone number in the United States. In this example, the **fax coverpage enable** command is superseded by the cover=no statement. No cover page will be generated.
`FAX=+1-312-555-3260/cover=yes@fax.com`	Fax sent to an E.164-compliant long distance telephone number in the United States. In this example, the **fax coverpage enable** command is superseded by the cover=yes statement. Store and Forward Fax will generate a fax coverpage.

Examples

The following example enables standard generated fax cover sheets:

```
configure terminal
 fax send coverpage enable
 fax send coverpage e-mail-controllable
```

Related Commands

Command	Description
fax send coverpage comment	Defines personalized text for the title field of a fax cover sheet.
fax send coverpage enable	Enables the Cisco AS5300 to generate fax cover sheets for faxes that originate from e-mail messages.
fax send coverpage show-detail	Prints all of the e-mail header information as part of the fax cover sheet.

fax send coverpage enable

To enable the Cisco AS5300 to generate fax cover sheets for faxes that originate from e-mail messages, use the **fax send coverpage enable** command in global configuration mode. Use the **no** form of this command to disable fax cover sheet generation functionality.

fax send coverpage enable

no fax send coverpage enable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Note This command is applicable only for faxes originating as e-mail messages. The Cisco AS5300 does not alter fax TIFF attachments. Therefore you cannot use this command to enable the AS5300 to generate fax cover pages for faxes being converted from TIFF files to standard fax transmissions.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example enables Cisco AS5300-generated fax cover sheets:

```
configure terminal
 fax send coverpage enable
```

Related Commands

Command	Description
fax send coverpage comment	Defines personalized text for the title field of a fax cover sheet.
fax send coverpage e-mail-controllable	Defers to the cover page setting in the e-mail header to generate a standard fax cover sheet.
fax send coverpage show-detail	Prints all of the e-mail header information as part of the fax cover sheet.

fax send coverpage show-detail

To print all of the e-mail header information as part of the fax cover sheet, use the **fax send coverpage show-detail** command in global configuration mode. Use the **no** form of this command to disable the e-mail header information being displayed.

fax send coverpage show-detail

no fax send coverpage show-detail

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command applies to off-ramp Store and Forward Fax functions.

Note This command is applicable only for faxes originating as e-mail messages. The Cisco AS5300 does not alter fax TIFF attachments. Therefore, you cannot use this command to enable the AS5300 to display additional fax cover page information for faxes being converted from TIFF files to standard fax transmissions.

Examples

The following example configures an individualized generated fax cover sheet that contains the e-mail header text:

```
configure terminal
 fax send coverpage enable
 no fax send coverpage e-mail-controllable
 fax send coverpage show-detail
```

Related Commands

Command	Description
fax send coverpage comment	Defines personalized text for the title field of a fax cover sheet.
fax send coverpage e-mail-controllable	Defers to the cover page setting in the e-mail header to generate a standard fax cover sheet.
fax send coverpage enable	Enables the Cisco AS5300 to generate fax cover sheets for faxes that originate from e-mail messages.

fax send left-header

To specify the data that will appear on the left in the fax header, use the **fax send left-header** command in global configuration mode. Use the **no** form of this command to disable the selected options.

fax send left-header {**a**| **d**| **p**| **s**| **t**| *string*}

no fax send left-header {**a**| **d**| **p**| **s**| **t**| *string*}

Syntax Description

a	Wildcard that inserts the date in the selected position.
d	Wildcard that inserts the destination address in the selected position.
p	Wildcard that inserts the page count in the selected position.
s	Wildcard that inserts the sender's address in the selected position.
t	Wildcard that inserts the transmission time in the selected position.
string	Text string that provides personalized information. Valid characters are any combination of ASCII characters and the wildcards listed.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Mail messages that contain only text or contain text attachments (MIME media type of text) can be converted by the off-ramp AS5300 into a format understood by fax machines using the AS5300's text-to-fax converter. When this conversion is performed, this **fax send left-header** command is used to indicate what header information should be added to the left top of those pages.

Mail messages with TIFF attachments (MIME media type of image and subtype of TIFF) are expected to include their own per-page headers and the AS5300's software does not modify TIFF attachments.

This command lets you configure multiple options at once—meaning that you can combine one or more wild cards with text string information to personalize your fax header information.

Note If the information you select for the **fax send left-header** command exceeds the space allocated for the left fax header, the information is truncated.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example selects the transmission time of the fax as the left fax header information:

```
configure terminal
 fax send left-header $t$
```

The following example configures the company name Widget and its address as the left fax header information:

```
configure terminal
 fax send left-header widget $s$
```

Related Commands

Command	Description
fax send center-header	Specifies the data that will appear in the center position of the fax header information.
fax send right-header	Specifies the data that will appear on the right in the fax header information.

fax send max-speed

To specify the maximum speed at which an outbound fax will be sent, use the **fax send max-speed** command in global configuration mode. Use the **no** form of this command to disable the selected speed.

fax send max-speed {**12000** | **14400** | **2400** | **4800** | **7200** | **9600**}

no fax send max-speed {**12000** | **14400** | **2400** | **4800** | **7200** | **9600**}

Syntax Description

12000	Indicates a transmission speed of 12000 bits per second (bps).
14400	Indicates a transmission speed of 14400 bps.
2400	Indicates a transmission speed of 2400 bps.
4800	Indicates a transmission speed of 4800 bps.
7200	Indicates a transmission speed of 7200 bps.
9600	Indicates a transmission speed of 9600 bps.

Defaults

14400

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example sets the outbound fax transmission rate at 2400 bps:

```
configure terminal
 fax send max-speed 2400
```

fax send right-header

To specify the data that will appear on the right in the fax header information, use the **fax send right-header** command in global configuration mode. Use the **no** form of this command to disable the selected options.

fax send right-header {**a**| **d**| **p**| **s**| **t**| *string*}

no fax send right-header {**a**| **d**| **p**| **s**| **t**| *string*}

Syntax Description

a	Wildcard that inserts the date in the selected position.
d	Wildcard that inserts the destination address in the selected position.
p	Wildcard that inserts the page count in the selected position.
s	Wildcard that inserts the sender's address in the selected position.
t	Wildcard that inserts the transmission time in the selected position.
string	Text string that provides personalized information. Valid characters are any combination of ASCII characters and the wildcards listed.

Defaults

No connection

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Mail messages that contain only text or contain text attachments (MIME media type of text) can be converted by the off-ramp AS5300 into a format understood by fax machines using the AS5300's text-to-fax converter. When this conversion is performed, this **fax send right-header** command is used to indicate what header information should be added to right top of those pages.

Mail messages with TIFF attachments (MIME media type of image and subtype of TIFF) are expected to include their own per-page headers and the AS5300's software does not modify TIFF attachments.

This command lets you configure multiple options at once—meaning that you can combine one or more wildcards with text string information to personalize your fax header information.

Note If the information you select for the **fax send right-header** command exceeds the space allocated for the right fax header, the information is truncated.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example selects the date of the fax as the right fax header information:

```
configure terminal
 fax send right-header $a$
```

The following example configures the company name Widget and its address as the right fax header information:

```
configure terminal
 fax send right-header widget $s$
```

Related Commands

Command	Description
fax send center-header	Specifies the data that will appear in the center position of the fax header information.
fax send left-header	Specifies the data that will appear on the left in the fax header.

fax send transmitting-subscriber

To define the transmitting subscriber identifier (TSI), use the **fax send transmitting-subscriber** command in global configuration mode. Use the **no** form of this command to disable the configured value.

fax send transmitting-subscriber {s| *string*}

no fax send transmitting-subscriber {s| *string*}

Syntax Description

s	Wildcard that inserts the sender name from the RFC 822 header (captured by the on-ramp from the sending fax machine) in the selected position.
string	Specifies the destination telephone number. Valid entries are the plus sign (+), numbers 0 through 9, and the space character. This string can specify an E.164 telephone number; if you choose to configure an E.164 telephone number, you must use the plus sign as the first character.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

The transmitting subscriber number is the number displayed in the LCD of the receiving fax device. Typically, with a standard Group 3 fax device, this is the telephone number associated with the transmitting or sending fax device. This command defines the TSI.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example configures the company name (as captured by the on-ramp from the sending fax machine as +18005551234) as the TSI:

```
configure terminal
 fax send transmitting-subscriber +14085551234
```

forward-digits

To configure forward digits for voice calls on the Cisco MC3810, use the **forward-digits** command in dial-peer configuration mode. Use the **no** form of this command to restore the default value.

forward-digits {*num-digit* | **all** | **implicit**}

no forward-digits {*num-digit* | **all** | **implicit**}

Syntax Description

num-digit	The number of digits to be forwarded. If the number of digits is longer than the length of a destination phone number, the length of the destination number is used. The valid range is 0 to 32. Setting the value to 0 is equivalent to entering **no forward digits** (the default).
all	Forward all digits. If "all" is used, the full length of the destination pattern will be used.
implicit	Exactly matched digits are not forwarded. Only digits matched by the wildcard pattern are forwarded.

Defaults

No digits are forwarded.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced.
12.0(2)T	The **implicit** option was added.
12.0(4)T	This command was modified to support ISDN PRI QSIG signalling calls.

Usage Guidelines

This command applies only to the Cisco MC3810.

This command applies only to POTS dial peers.

Forwarded digits are always right-justified so that extra leading digits are stripped.

For QSIG ISDN connections, entering the **forward-digits all** command implies that all of the digits of the called party number are sent to the ISDN connection. When you enter the **forward-digits** *num-digit* command and enter a number between 1 and 32, the number of digits specified (right justified) of the called part number are sent to the ISDN connection. When you enter the **forward-digits implicit** command, then the number of matched digits matching the wildcard for the called party number are sent to the ISDN connection. For example, if the called part number is 51234, the digits being sent to the ISDN connection is 1234.

Examples

The following example configures forward digits for a POTS dial peer on a Cisco MC3810:

```
dial-peer voice 1 pots
 destination-pattern 8...
 forward-digits all
```

Related Commands

Command	Description
destination pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.

frame-relay voice bandwidth

To specify how much bandwidth should be reserved for voice traffic on a specific data-link connection identifier (DLCI), use the **frame-relay voice bandwidth** map-class configuration command. Use the **no** form of this command to release the bandwidth previously reserved for voice traffic.

frame-relay voice bandwidth *bps_reserved* [**queue** *depth*]

no frame-relay voice bandwidth *bps_reserved* [**queue** *depth*]

Syntax Description

bps_reserved	The bandwidth in bps reserved for voice traffic for the specified map class. The range is from 8000 to 45000000 bits per second; the default is 0, which disables voice calls.
queue *depth*	(Optional) The queue reserved strictly for voice packets.The *depth* value represents the depth of the queue reserved strictly for voice packets. The default is 100, and the valid range is 30 to 1000.

Defaults

Disabled (zero)

Command Modes

Map-class configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.
12.0(5)T	The **queue** *depth* keyword and argument were added.

Usage Guidelines

To use this command, you must first associate a Frame Relay map class with a specific DLCI, then enter map-class configuration mode and set the amount of bandwidth to be reserved for voice traffic for that map class.

If a call is attempted and there is not enough remaining bandwidth reserved for voice to handle the additional call, the call will be rejected. For example, if 64 kbps is reserved for voice traffic, and a codec and payload size is being used that requires 10 kbps of bandwidth for each call, then the first six calls attempted will be accepted, but the seventh call will be rejected.

Reserve queues are not required for Voice over Frame Relay.

Note Cisco strongly recommends that you set voice bandwidth to a value less than the committed information rate (CIR) if Frame Relay traffic shaping is configured. Cisco also strongly recommends that you set the minimum CIR (using the **frame-relay mincir** command) to be at least equal to or greater than the voice bandwidth.

When you set the **queue** *depth* option, keep the depth small. Queueing packets on the voice queue indicates that there is some congestion on the PVC. Queueing too many packets on this queue indicates that there are more voice calls allowed on this PVC than it can handle. In this situation, it is recommended that you decrease the number of calls allowed on the PVC. Note that heavy data congestion may cause some voice packets to be queued, but given the priority of servicing the voice queue, the congestion will not cause the voice queue to be too deep.

Calculating Required Bandwidth

The bandwidth required for a voice call depends on the bandwidth of the codec, the voice packetization overhead, and the voice frame payload size. The smaller the voice frame payload size, the higher the bandwidth required for the call. To make the calculation, use the following formula:

required_bandwidth = codec_bandwidth x (1 + overhead / payload_size)

As an example, the overhead for VoFR voice packet is between 6 and 8 bytes: a 2-byte Frame Relay header, a 1- or 2-byte FRF.11 header (depending on the CID value), a 2-byte CRC, and a 1-byte trailing flag. If voice sequence numbers are enabled in the voice packets, there is an additional 1-byte sequence number. Table 13 shows the required voice bandwidth for the G.729 8000 bps speech coder for various payload sizes.

Table 13 Required Voice Bandwidth Calculations for G.729

Codec	Codec Bandwidth	Voice Frame Payload Size	Required Bandwidth per Call (6-Byte OH)	Required Bandwidth per Call (8-Byte OH)
G.729	8000 bps	120 bytes	8400 bps	8534 bps
G.729	8000 bps	80 bytes	8600 bps	8800 bps
G.729	8000 bps	40 bytes	9200 bps	9600 bps
G.729	8000 bps	30 bytes	9600 bps	10134 bps
G.729	8000 bps	20 bytes	10400 bps	11200 bps

To configure the payload size for the voice frames, use the **codec** command from dial-peer configuration mode.

Examples

The following example shows how to reserve 64 kbps for voice traffic for the "vofr" Frame Relay map class on a Cisco 2600 series, 3600 series, or 7200 series router or on an MC3810 concentrator:

```
interface serial 1/1
 frame-relay interface-dlci 100
  class vofr
  exit
map-class frame-relay vofr
 frame-relay voice bandwidth 64000
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
frame-relay fair-queue	Enables weighted fair queueing for one or more Frame Relay PVCs.
frame-relay fragment	Enables fragmentation for a Frame Relay map class.
frame-relay interface-dlci	Assigns a DLCI to a specified Frame Relay subinterface on the router or access server.
map-class frame-relay	Specifies a map class to define QoS values for an SVC.

ftc-trunk frame-relay-dlci

To map the FTC Frame Relay data-link connection identifier (DLCI) to an existing Frame Relay DLCI for data traffic on the Cisco MC3810, use the **ftc-trunk frame-relay-dlci** command in interface configuration mode. To remove the FTC Frame Relay DLCI from the interface, use the **no** form of this command.

ftc-trunk frame-relay-dlci *dlci* [**remote-frame-relay-dlci** *dlci*]

no ftc-trunk frame-relay-dlci *dlci* [**remote-frame-relay-dlci** *dlci*]

Syntax Description

dlci	Specifies the Frame Relay DLCI for the IGX switch.
remote-frame-relay-dlci *dlci*	(Optional) Specifies the remote Frame Relay DLCI if there is a public Frame Relay cloud between the Cisco MC3810 FTC port and the IGX FTM port. Enter this option only if the local DLCI and the remote DLCI are not the same.

Defaults

No FTC trunk Frame Relay DLCI is configured.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(2)T	This command was introduced.

Usage Guidelines

Do not enter the same DLCI that was assigned to the FTC management DLCI using the **ftc-trunk management-dlci** command.

Examples

The following configures DLCI 200 as the FTC trunk Frame Relay DLCI:

```
interface serial 1
 ftc-trunk frame-relay-dlci 200
```

Related Commands

Command	Description
connect voice	Configures a connection on the Cisco MC3810 multiservice concentrator between a voice dial peer and the FTC trunk.
encapsulation ftc-trunk	Enables FTC trunk encapsulation on a serial interface on the Cisco MC3810 multiservice concentrator.
ftc-trunk management-dlci	Maps the FTC management DLCI to an existing Frame Relay DLCI for management traffic on the Cisco MC3810.

Command	Description
ftc-trunk management-protocol	Selects the mode on the Cisco MC3810 that management frames are sent on the FTC trunk management DLCI.
ftc-trunk voice-dlci	Maps the FTC voice DLCI to an existing Frame Relay DLCI for voice traffic on the Cisco MC3810.

ftc-trunk management-dlci

To map the FTC management data-link connection identifier (DLCI) to an existing Frame Relay DLCI for management traffic on the Cisco MC3810, use the **ftc-trunk management-dlci** command in interface configuration mode. To remove the FTC management DLCI from the interface, use the **no** form of this command.

ftc-trunk management-dlci *dlci* [**remote-management-dlci** *dlci*]

no ftc-trunk management-dlci *dlci* [**remote-management-dlci** *dlci*]

Syntax Description

dlci	Specifies the Frame Relay DLCI for the IGX switch.
remote-management-dlci *dlci*	(Optional) Specifies the remote management DLCI if there is a public Frame Relay cloud between the Cisco MC3810 FTC port and the IGX FTM port. Enter this option only if the local DLCI and the remote DLCI are not the same.

Defaults

No FTC trunk management DLCI is configured.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(2)T	This command was introduced.

Usage Guidelines

The DLCI (or the remote DLCI if configured) must match the DLCI on the IGX switch configured using the IGX **addad** command.

Examples

The following configures DLCI 100 as the FTC trunk management DLCI:

```
interface serial 1
 encapsulation ftc-trunk
 ftc-trunk management-dlci 100
```

Related Commands

Command	Description
connect (global)	Configures a data connection on the Cisco MC3810 to a 16x switch that will travel over the FTC trunk.
connect voice	Configures a connection on the Cisco MC3810 multiservice concentrator between a voice dial peer and the FTC trunk.
encapsulation ftc-trunk	Enables FTC trunk encapsulation on a serial interface on the Cisco MC3810 multiservice concentrator.

Command	Description
ftc-trunk frame-relay-dlci	Maps the FTC Frame Relay DLCI to an existing Frame Relay DLCI for data traffic on the Cisco MC3810.
ftc-trunk management-protocol	Selects the mode on the Cisco MC3810 that management frames are sent on the FTC trunk management DLCI.
ftc-trunk voice-dlci	Maps the FTC voice DLCI to an existing Frame Relay DLCI for voice traffic on the Cisco MC3810.

ftc-trunk management-protocol

To select the mode on the Cisco MC3810 that management frames are sent on the FTC trunk management data-link connection identifier (DLCI), use the **ftc-trunk management-protocol** command in interface configuration mode. To restore the default value, use the **no** form of this command.

ftc-trunk management-protocol {**normal** | **inverted**}

no ftc-trunk management-protocol {**normal** | **inverted**}

Syntax Description

normal	Specifies normal mode.
inverted	Specifies inverted mode.

Defaults

Normal mode

Command Modes

Interface configuration

Command History

Release	Modification
12.0(2)T	This command was introduced.

Usage Guidelines

Use **normal** mode when the Cisco MC3810 is connected directly to the IGX switch. When two Cisco MC3810 concentrators are connected back-to-back, use **inverted** mode on one concentrator and **normal** mode on the other concentrator.

This command is available only when the **ftc-trunk management-dlci** command is configured.

Examples

The following configures the FTC trunk management protocol to normal mode:

```
interface serial 1
 encapsulation ftc-trunk
 ftc-trunk management-dlci 100
 ftc-trunk management-protocol normal
```

Related Commands

Command	Description
connect (global)	Configures a data connection on the Cisco MC3810 to a 16x switch that will travel over the FTC trunk.
connect voice	Configures a connection on the Cisco MC3810 multiservice concentrator between a voice dial peer and the FTC trunk.
encapsulation ftc-trunk	Enables FTC trunk encapsulation on a serial interface on the Cisco MC3810 multiservice concentrator.
ftc-trunk frame-relay-dlci	Maps the FTC Frame Relay DLCI to an existing Frame Relay DLCI for data traffic on the Cisco MC3810.

Command	Description
ftc-trunk management-dlci	Maps the FTC management DLCI to an existing Frame Relay DLCI for management traffic on the Cisco MC3810.
ftc-trunk voice-dlci	Maps the FTC voice DLCI to an existing Frame Relay DLCI for voice traffic on the Cisco MC3810.

ftc-trunk voice-dlci

To map the FTC voice data-link connection identifier (DLCI) to an existing Frame Relay DLCI for voice traffic on the Cisco MC3810, use the **ftc-trunk voice-dlci** command in interface configuration mode. To remove the FTC voice DLCI from the interface, use the **no** form of this command.

ftc-trunk voice-dlci *dlci* [**remote-voice-dlci** *dlci*]

no ftc-trunk voice-dlci *dlci* [**remote-voice-dlci** *dlci*]

Syntax Description

dlci	Specifies the Frame Relay DLCI for the IGX switch.
remote-voice-dlci *dlci*	(Optional) Specifies the remote voice DLCI If there is a public Frame Relay cloud between the Cisco MC3810 FTC port and the IGX FTM port. Enter this option only if the local DLCI and the remote DLCI are not the same.

Defaults

No FTC trunk voice DLCI is configured.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(2)T	This command was introduced.

Usage Guidelines

Do not enter the same DLCI that was assigned to the FTC management DLCI using the **ftc-trunk management-dlci** command.

Examples

The following configures the FTC trunk voice DLCI:

```
interface serial 1
 ftc-trunk voice-dlci 100
```

Related Commands

Command	Description
connect (global)	Configures a data connection on the Cisco MC3810 to a 16x switch that will travel over the FTC trunk.
connect voice	Configures a connection on the Cisco MC3810 multiservice concentrator between a voice dial peer and the FTC trunk.
encapsulation ftc-trunk	Enables FTC trunk encapsulation on a serial interface on the Cisco MC3810 multiservice concentrator.
ftc-trunk frame-relay-dlci	Maps the FTC Frame Relay DLCI to an existing Frame Relay DLCI for data traffic on the Cisco MC3810.

Command	Description
ftc-trunk management-dlci	Maps the FTC management DLCI to an existing Frame Relay DLCI for management traffic on the Cisco MC3810.
ftc-trunk management-protocol	Selects the mode on the Cisco MC3810 that management frames are sent on the FTC trunk management DLCI.

gatekeeper

To enter gatekeeper configuration mode, use the **gatekeeper** command in global configuration mode.

gatekeeper

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Usage Guidelines

Press **Ctrl-Z** or use the **exit** command to exit gatekeeper configuration mode.

Examples

The following example brings the gatekeeper online:

```
configure terminal
gatekeeper
no shutdown
```

gateway

To enable the H.323 Voice over IP gateway, use the **gateway** command in global configuration mode. Use the **no** form of this command to unregister this gateway with the gatekeeper.

gateway

no gateway

Syntax Description

This command has no keywords or arguments.

Defaults

The gateway is unregistered.

Command Modes

Global configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Usage Guidelines

Use the **gateway** command to enable H.323 VoIP gateway functionality. After you enable the gateway, it will attempt to discover a gatekeeper by using the H.323 RAS GRQ message. If you enter **no gateway voip**, the VoIP gateway will unregister with the gatekeeper via the H.323 RAS URQ message.

Examples

The following example enables the gateway:

```
configure terminal
gateway
```

gw-accounting

To enable gateway-specific accounting, use the **gw-accounting** command in global configuration mode. Use the **no** form of this command to disable gateway specific accounting.

gw-accounting {**h323** | **syslog** | **vsa**}

no gw-accounting {**h323** | **syslog** | **vsa**}

Syntax Description

h323	Enables standard H.323 accounting using Internet Engineering task Force (IETF) RADIUS attributes.
syslog	Enables the system logging facility to output accounting information in the form of a system log message.
vsa	Enables H.323 accounting using RADIUS vendor specific attributes (VSAs).

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced.
12.0(7)T	The **vsa** field was added to this command.

Usage Guidelines

To collect basic start-stop connection accounting data, the gateway must be configured to support gateway-specific H.323 accounting functionality. The **gw-accounting** command enables you to send accounting data to the RADIUS server in one of four ways:

- Using standard IETF RADIUS accounting attribute/value (AV) pairs—This method is the basic method of gathering accounting data (connection accounting) per the specifications defined by the IETF. Use the **gw-accounting h323** command to configure the standard IETF RADIUS method of applying H.323 gateway-specific accounting. Table 14 shows the IETF RADIUS attributes supported.

Table 14 Supported IETF RADIUS Accounting Attributes

Number	Attributes	Description
30	Called-Station-Id	(Accounting) Allows the network access server to send the telephone number the user called as part of the Access-Request packet (using Dialed Number Identification [DNIS] or similar technology). This attribute is only supported on ISDN, and modem calls on the Cisco AS5200 and Cisco AS5300 if used with ISDN PRI.

Table 14 Supported IETF RADIUS Accounting Attributes (continued)

Number	Attributes	Description
31	Calling-Station-Id	(Accounting) Allows the network access server to send the telephone number the call came from as part of the Access-Request packet (using Automatic Number Identification or similar technology). This attribute has the same value as "remote-addr" from TACACS+. This attribute is only supported on ISDN, and modem calls on the Cisco AS5200 and Cisco AS5300 if used with ISDN PRI.
42	Acct-Input-Octets	(Accounting) Indicates how many octets have been received from the port over the course of the accounting service being provided.
43	Acct-Output-Octets	(Accounting) Indicates how many octets have been sent to the port over the course of delivering the accounting service.
44	Acct-Session-Id	(Accounting) Indicates a unique accounting identifier that makes it easy to match start and stop records in a log file. Acct-Session-Id numbers restart at 1 each time the router is power-cycled or the software reloaded.
47	Acct-Input-Packets	(Accounting) Indicates how many packets have been received from the port over the course of this service being provided to a framed user.
48	Acct-Output-Packets	(Accounting) Indicates how many packets have been sent to the port in the course of delivering this service to a framed user.

For more information about RADIUS and the use of IETF-defined attributes, refer to the *Cisco IOS Security Configuration Guide.*

- Overloading the Acct-Session-Id field—Attributes that cannot be mapped to standard RADIUS are packed into the Acct-Session-Id attribute field as ASCII strings separated by the character "/". The Acct-Session-Id attribute is defined to contain the RADIUS account session ID, which is a unique identifier that links accounting records associated with the same login session for a user. To support additional fields, we have defined the following string format for this field:

```
<session id>/<call leg setup time>/<gateway id>/<connection id>/<call origin>/
<call type>/<connect time>/disconnect time>/<disconnect cause>/<remote ip
address>
```

Table 15 shows the field attributes that you use with the Overloaded Session-ID method and a brief description of each.

Table 15 Field Attributes in Overloaded Acct-Session ID

Field Attributes	Description
Session-Id	Specifies the standard RADIUS account session ID.
Setup-Time	Provides the Q.931 setup time for this connection in Network Time Protocol (NTP) format. NTP time formats are displayed as%H: %M: %S %k %Z %tw %tn %td %Y where: %H is hour (00 to 23) %M is minutes (00 to 59) %S is seconds (00 to 59) %k is milliseconds (000 to 999) %Z is timezone string %tw is day of week (Saturday through Sunday) %tn is month name (January through December) %td is day of month (01 to 31) %Y is year including century (for example, 1998)
Gateway-Id	Indicates the name of the underlying gateway in the form of "gateway.domain_name."
Call-Origin	Indicates the origin of the call relative to the gateway. Possible values are **originate** and **answer**.
Call-Type	Indicates call leg type. Possible values are **telephony** and **VoIP**.
Connection-Id	Specifies the unique global identifier used to correlate call legs that belong to the same end-to end call. The field consists of 4 long words (128 bits). Each long word is displayed as a hexadecimal value and separated by a space character.
Connect-Time	Provides the Q.931 connect time for this call leg, in NTP format.
Disconnect-Time	Provides the Q.931 disconnect time for this call leg, in NTP format.
Disconnect-Cause	Specifies the reason a call was taken off-line as defined in the Q.931 specification.
Remote-Ip-Address	Indicates the address of the remote gateway port where the call is connected.

Because of the limited size of the Acct Session Id string, it is not possible to embed very many information elements in it. Therefore, this feature supports only a limited set of accounting information elements.

Use the **gw-accounting h323** command to configure the overloaded session ID method of applying H.323 gateway-specific accounting.

- Using vendor-specific RADIUS attributes—The IETF draft standard specifies a method for communicating vendor-specific information between the network access server and the RADIUS server by using the vendor-specific attribute (Attribute 26). Vendor-specific attributes (VSAs) allow vendors to support their own extended attributes not suitable for general use. The Cisco RADIUS implementation supports one vendor-specific option using the format recommended in the specification. Cisco's vendor-ID is 9, and the supported option has vendor-type 1, which is named "cisco-avpair." The value is a string of the format:

```
protocol: attribute sep value *
```

  "Protocol" is a value of the Cisco "protocol" attribute for a particular type of authorization. "Attribute" and "value" are an appropriate attribute/value (AV) pair defined in the Cisco TACACS+ specification, and "sep" is "=" for mandatory attributes and "*" for optional attributes. This allows the full set of features available for TACACS+ authorization to also be used for RADIUS.

  The VSA fields and their ASCII values are listed in Table 16.

Table 16 VSA Fields and Their ASCII Values

IETF RADIUS Attribute	Vendor-Specific Company Code	Subtype Number	Attribute Name	Description
26	9	23	h323-remote-address	Indicates the IP address of the remote gateway.
26	9	24	h323-conf-id	Identifies the conference ID.
26	9	25	h323-setup-time	Indicates the setup time for this connection in Coordinated Universal Time (UTC) formerly known as Greenwich Mean Time (GMT) and Zulu time.
26	9	26	h323-call-origin	Indicates the origin of the call relative to the gateway. Possible values are originating and terminating (answer).
26	9	27	h323-call-type	Indicates call leg type. Possible values are **telephony** and **VoIP**.
26	9	28	h323-connect-time	Indicates the connection time for this call leg in UTC.
26	9	29	h323-disconnect-time	Indicates the time this call leg was disconnected in UTC.
26	9	30	h323-disconnect-cause	Specifies the reason a connection was taken off-line per Q.931 specification.
26	9	31	h323-voice-quality	Specifies the impairment factor (ICPIF) affecting voice quality for a call.
26	9	33	h323-gw-id	Indicates the name of the underlying gateway.

Use the **gw-accounting h323 vsa** command to configure the VSA method of applying H.323 gateway-specific accounting.

- Using syslog records—The syslog accounting option exports the information elements associated with each call leg through a system log message, which can be captured by a syslog daemon on the network. The syslog output consists of the following:

```
<server timestamp> <gateway id> <message number> : <message label> : <list of AV
pairs>
```

The syslog messages fields are listed in Table 17.

Table 17 Syslog Message Output Fields

Field	Description
server timestamp	The time stamp created by the server when it receives the message to log.
gateway id	The name of the gateway emitting the message.
message number	The number assigned to the message by the gateway.
message label	Is a string used to identify the message category.
list of AV pairs	Is a string consisting of <attribute name> <attribute value> pairs separated by commas.

Use the **gw-accounting h323 syslog** command to configure the syslog record method of gathering H.323 accounting data.

Examples

The following example configures the basic H.323 accounting using IETF RADIUS attributes:

```
gw-accounting h323
```

The following example configures H.323 accounting using the VSA RADIUS attributes:

```
gw-accounting vsa
```

gw-type-prefix

To configure a technology prefix in the gatekeeper, use the **gw-type-prefix** command in gatekeeper configuration mode. To remove the technology prefix, use the **no** form of the command.

gw-type-prefix *type-prefix* [**hopoff** *gkid*] [**default-technology**] [**gw ipaddr** *ipaddr* [*port*]]

no gw-type-prefix *type-prefix* [**hopoff** *gkid*] [**default-technology**] [**gw ipaddr** *ipaddr* [*port*]]

Syntax Description

type-prefix	A technology prefix is recognized and is stripped before checking for the zone prefix. It is strongly recommended that you select technology prefixes that do not lead to ambiguity with zone prefixes. Do this by using the # character to terminate technology prefixes, for example, 3#.
hopoff *gkid*	(Optional) Specifies the gatekeeper or zone where the call is to hop off, regardless of the zone prefix in the destination address. The *gkid* argument refers to a zone previously configured using the zone local or zone remote comment.
default-technology	(Optional) Gateways registering with this prefix option are used as the default for routing any addresses that are otherwise unresolved.
gw ipaddr *ipaddr* [*port*]	(Optional) Indicates that the gateway is incapable of registering technology prefixes. When it registers, it adds the gateway to the group for this type-prefix, just as if it had sent the technology prefix in its registration. This parameter can be repeated to associate more than one gateway with a technology prefix.

Defaults

No technology prefix is defined.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Usage Guidelines

More than one gateway can register with the same technology prefix. In such cases, a random selection is made of one of them.

You do not have to define a technology prefix to a gatekeeper if there are gateways configured to register with that prefix, and if there are no special flags (**hopoff** *gkid* or **default-technology**) that you want to associate with that prefix.

You need to configure the gateway type prefix of all remote technology prefixes that will be routed through this gatekeeper.

Examples

The following example specifies 4# as the default technology prefix:

```
default-technology
```

Related Commands

Command	Description
zone prefix	Configures the gatekeeper with knowledge of its own prefix and the prefix of any remote zone.

h323 asr

To enable application-specific routing (ASR), use the **h323 asr** command in interface configuration mode. Use the **no** form of this command to disable ASR.

h323 asr

no h323 asr

Syntax Description

There are no arguments or keywords for this command.

Defaults

ASR is disabled.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced.

Usage Guidelines

This command is independent of the **h323 interface** command.

This command is not supported on Frame Relay or ATM interfaces for the Cisco MC3810 platform.

Note If you specify **no h323 asr bandwidth** *max-bandwidth*, this removes the bandwidth setting but ASR is still enabled. You must enter **no h323 asr** to disable ASR.

Examples

The following example enables ASR:

```
h323 asr
```

h323 asr bandwidth

To specify the maximum bandwidth for a proxy, use the **h323 asr bandwidth** command in interface configuration mode. Use **no** form of this command to remove a bandwidth setting but keep ASR enabled.

h323 asr [**bandwidth** *max-bandwidth*]

no h323 asr [**bandwidth** *max-bandwidth*]

Syntax Description

bandwidth *max-bandwidth*	(Optional) Maximum bandwidth on the interface. Value ranges from 1 to 10,000,000 kbps. If you do not specify the *max-bandwidth*, this value defaults to the bandwidth on the interface. If you specify *max-bandwidth* as a value greater than the interface bandwidth, the bandwidth will default to the interface bandwidth.

Defaults

ASR is disabled.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Usage Guidelines

This command is independent of the **h323 interface** command.

This command is not supported on Frame Relay or ATM interfaces for the Cisco MC3810 platform.

Note If you specify **no h323 asr bandwidth** *max-bandwidth*, this removes the bandwidth setting and ASR is still enabled. You must enter **no h323 asr** to disable ASR.

Examples

The following example enables ASR and specifies a maximum bandwidth of 10,000 kbps:

```
h323 asr bandwidth 10000
```

h323 gatekeeper

To specify the gatekeeper associated with aproxy and control how the gatekeeper is discovered, use the **h323 gatekeeper** command in interface configuration mode. Use the **no** form of this command to disassociate the gatekeeper.

h323 gatekeeper [**id** *gatekeeper-id]* {**ipaddr** *ipaddr* [*port*] | **multicast**}

no h323 gatekeeper [**id** *gatekeeper-id*] {**ipaddr** *ipaddr* [*port*] | **multicast**}

Syntax Description

id *gatekeeper-id*	(Optional) The *gatekeeper-id* argument specifies the gatekeeper name. Typically, this is a DNS name, but it can also be a raw IP address in dotted form. If this parameter is specified, gatekeepers that have either the default or explicit flags set for the proxy's subnet will respond. If this parameter is not specified, only those gatekeepers with the default subnet flag will respond.
ipaddr *ipaddr* [*port*]	If this parameter is specified, the gatekeeper discovery message will be unicast to this address and, optionally, the port specified.
multicast	If this parameter is specified, the gatekeeper discovery message will be multicast to the well-known RAS multicast address and port.

Defaults

No gatekeeper is configured for the proxy.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced.

Usage Guidelines

You must enter the **h323 interface** and **h323 h323-id** commands before using this command. The **h323 gatekeeper** command must be specified on your Cisco IOS platform or the proxy will not go online. The proxy will use the interface's address as its RAS signalling address.

Examples

The following example sets up a unicast discovery to a gatekeeper whose name is unknown:

```
h323 gatekeeper ipaddr 191.7.5.2
```

The following example sets up a multicast discovery for a gatekeeper of a particular name:

```
h323 gatekeeper id gk.zone5.com multicast
```

Related Commands

Command	Description
h323 interface	Registers an H.323 proxy alias with a gatekeeper.
h323 interface	Specifies the interface from which the proxy will take its IP address.

h323-gateway voip h323-id

To configure the H.323 name of the gateway identifying this gateway to its associated gatekeeper, use the **h323-gateway voip h.323-id** command in interface configuration mode. Use the **no** form of this command to disable this defined gatekeeper name.

h323-gateway voip h323-id *interface-id*

no h323-gateway voip h323-id *interface-id*

Syntax Description

interface-id	H.323 name (ID) used by this gateway when this gateway communicates with its associated gatekeeper. Usually, this ID is the name of the gateway with the gatekeeper domain name appended to the end: name@domain-name.

Defaults

No gateway identification is defined.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Examples

The following example configures Ethernet interface 0.0 as the gateway interface. In this example, the gateway ID is GW13@cisco.com.

```
interface Ethernet0/0
 ip address 172.9.53.13 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id GK15.cisco.com ipaddr 172.9.53.15 1719
 h323-gateway voip h323-id GW13@cisco.com
 h323-gateway voip tech-prefix 13#
```

Related Commands

Command	Description
h323-gateway voip id	Defines the name and location of the gatekeeper for this gateway.
h323-gateway voip interface	Configures this interface as an H.323 interface.
h323-gateway voip tech-prefix	Defines the technology prefix that the gateway will register with the gatekeeper.

h323-gateway voip id

To define the name and location of the gatekeeper for a specific gateway, use the **h323-gateway voip id** command in interface configuration mode. Use the **no** form of this command to disable this gatekeeper identification.

h323-gateway voip id *gatekeeper-id* {**ipaddr** *ip-address* [*port-number*] | **multicast**}

no h323-gateway voip id *gatekeeper-id* {**ipaddr** *ip-address* [*port-number*] | **multicast**}

Syntax Description

gatekeeper-id	Indicates the H.323 identification of the gatekeeper. This value must exactly match the gatekeeper ID in the gatekeeper configuration. The recommended format is *name.doman-name*.
ipaddr	Indicates that the gateway will use an IP address to locate the gatekeeper
ip-address	Defines the IP address used to identify the gatekeeper.
multicast	Indicates that the gateway will use multicast to locate the gatekeeper.
port-number	(Optional) Defines the port number used.

Defaults

No gatekeeper identification is defined.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Usage Guidelines

This command tells the H.323 gateway associated with this interface which H.323 gatekeeper to talk to and where to locate it. The gatekeeper ID configured here must exactly match the gatekeeper ID in the gatekeeper configuration.

Examples

The following example configures Ethernet interface 0.0 as the gateway interface. In this example, the gatekeeper ID is GW15.cisco.com and its IP address is 172.9.53.15 (using port 1719).

```
interface Ethernet0/0
 ip address 172.9.53.15 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id GK15.cisco.com ipaddr 172.9.53.15 1719
 h323-gateway voip h323-id GW15@cisco.com
 h323-gateway voip tech-prefix 13#
```

Related Commands

Command	Description
h323-gateway voip h323-id	Configures the H.323 name of the gateway identifying this gateway to its associated gatekeeper.
h323-gateway voip interface	Configures this interface as an H.323 interface.
h323-gateway voip tech-prefix	Defines the technology prefix that the gateway will register with the gatekeeper.

h323-gateway voip interface

To configure this interface as an H.323 gateway interface, use the **h323-gateway voip interface** command in interface configuration mode. Use the **no** form of this command to disable H.323 functionality for this interface.

h323-gateway voip interface

no h323-gateway voip interface

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Examples

The following example configures Ethernet interface 0.0 as the gateway interface. In this example, the **h323-gateway voip interface** command configures this interface as an H.323 interface.

```
interface Ethernet0/0
 ip address 172.9.53.15 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id GK15.cisco.com ipaddr 172.9.53.15 1719
 h323-gateway voip h323-id GW15@cisco.com
 h323-gateway voip tech-prefix 13#
```

Related Commands

Command	Description
h323-gateway voip h323-id	Configures the H.323 name of the gateway identifying this gateway to its associated gatekeeper.
h323-gateway voip id	Defines the name and location of the gatekeeper for this gateway.
h323-gateway voip tech-prefix	Defines the technology prefix that the gateway will register with the gatekeeper.

h323-gateway voip tech-prefix

To define the technology prefix that the gateway will register with the gatekeeper, use the **h323-gateway voip tech-prefix** command in interface configuration mode. Use the **no** form of this command to disable this defined technology prefix.

h323-gateway voip tech-prefix *prefix*

no h323-gateway voip tech-prefix *prefix*

Syntax Description

prefix	Defines the numbers used as the technology prefixes. Each technology prefix can contain up to 11 characters. Although not strictly necessary, a pound (#) symbol is frequently used as the last digit in a technology prefix. Valid characters are 0 though 9, the pound (#) symbol, and the asterisk (*).

Defaults

Disabled

Command Modes

Interface configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Usage Guidelines

This command defines a technology prefix that the gateway will then register with the gatekeeper. Technology prefixes can be used as a discriminator so that the gateway can tell the gatekeeper that a certain technology is associated with a particular call (for example, 15# could mean a fax transmission), or it can be used like an area code for more generic routing. No standard currently defines what the numbers in a technology prefix mean. By convention, technology prefixes are designated by a pound (#) symbol as the last character.

Note Cisco gatekeepers use the asterisk (*) as a reserved character. If you are using Cisco gatekeepers, do not use the asterisk as part of the technology prefix.

Examples

The following example configures Ethernet interface 0.0 as the gateway interface. In this example, the technology prefix is defined as 13#.

```
interface Ethernet0/0
 ip address 172.9.53.15 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id GK15.cisco.com ipaddr 172.9.53.15 1719
 h323-gateway voip h323-id GW15@cisco.com
 h323-gateway voip tech-prefix 13#
```

Related Commands

Command	Description
h323-gateway voip id	Defines the name and location of the gatekeeper for this gateway.
h323-gateway voip interface	Configures this interface as an H.323 interface.
h323-gateway voip h323-id	Configures the H.323 name of the gateway identifying this gateway to its associated gatekeeper.

h323 h323-id

To register an H.323 proxy alias with a gatekeeper, use the **h323 h323-id** command in interface configuration mode. To remove an H.323 alias, use the **no** form of this command.

h323 h323-id *h323-id*

no h323 h323-id *h323-id*

Syntax Description

h323-id	Specifies the name of the proxy. It is recommended that this be a fully qualified e-mail ID, with the domain name being the same as that of its gatekeeper.

Defaults

No *h323-id* proxy alias is registered.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Usage Guidelines

Each entry registers a specified H.323 ID proxy alias to a gatekeeper. Typically, these aliases are either simple text strings or legitimate e-mail IDs.

Note You must enter the **h323 interface** command before using this command. The **h323 h323-id** command must be entered on the same interface as the **h323 gatekeeper** command. The proxy will not go online without this command.

Examples

The following example registers an H.323 proxy alias called proxy1@zone5.com with a gatekeeper:

```
h323 h323-id proxy1@zone5.com
```

Related Commands

Command	Description
h323 gatekeeper	Specifies the gatekeeper associated with a proxy and controls how the gatekeeper is discovered.
h323 interface	Specifies the interface from which the proxy will take its IP address.

h323 interface

To specify the interface from which the proxy will take its IP address, use the **h323 interface** command in interface configuration mode. To disable the interface, use the **no** form of this command.

h323 interface

no h323 interface

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Usage Guidelines

For non-ASR configurations, any interface on the Cisco IOS platform will work well as the proxy interface. For ASR configurations, the proxy interface should be a loopback interface, so that routing updates and packet switching are appropriately isolated between the ASR and non-ASR interfaces.

Examples

The following example specifies the interface from which the proxy will take its IP address:

```
interface Loopback0
 ip address 173.0.0.1 255.0.0.0
 h323 interface
```

Related Commands

Command	Description
h323 qos	Enables QoS on the proxy.

h323 qos

To enable quality of service (QoS) on the proxy, use the **h323 qos** command in interface configuration mode. To disable QoS, use the **no** form of this command.

h323 qos {**ip-precedence** *value* | **rsvp** {**controlled-load** | **guaranteed-qos**}}

no h323 qos {**ip-precedence** *value* | **rsvp** {**controlled-load** | **guaranteed-qos**}}

Syntax Description

ip-precedence *value*	Specifies that RTP streams should set their IP precedence bits to the specified *value*.
rsvp controlled-load	Specifies controlled load class of service.
rsvp guaranteed-qos	Specifies guaranteed QoS class of service.

Defaults

No QoS is configured.

Command Modes

Interface configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced.

Usage Guidelines

You must execute the **h323 interface** command before using this command.

Both IP precedence and RSVP QoS can be configured by invoking this command twice with the two different QoS forms.

Examples

The following example enables QoS on the proxy:

```
interface Ethernet0
 ip address 172.21.127.38 255.255.255.192
 no ip redirects
 ip rsvp bandwidth 7000 7000
 ip route-cache same-interface
 fair-queue 64 256 1000
 h323 interface
 h323 qos rsvp controlled-load
 h323 h323-id px1@zone1.com
 h323 gatekeeper ipaddr 172.21.127.39
```

Related Commands

Command	Description
h323 interface	Specifies the interface from which the proxy will take its IP address.

huntstop

To disable all further dial-peer hunting if a call fails when using hunt groups, enter the **huntstop** dial-peer configuration command. To reenable dial peer call hunting, enter the **no** form of this command.

huntstop

no huntstop

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco MC3810.

Usage Guidelines

Once you enter this command, no further hunting is allowed if a call fails on the specified dial peer.

Note This command can be used with all types of dial peers.

Examples

The following example shows how to disable dial-peer hunting on a specific dial peer:

```
dial peer voice 100 vofr
 huntstop
```

The following example shows how to reenable dial-peer hunting on a specific dial peer:

```
dial peer voice 100 vofr
 no huntstop
```

Related Commands

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

icpif

To specify the Calculated Planning Impairment Factor (ICPIF) for calls sent by a dial peer, use the **icpif** command in dial-peer configuration mode. Use the **no** form of this command to restore the default value.

icpif *number*

no icpif *number*

Syntax Description

number	Integer, expressed in equipment impairment factor units, specifying the ICPIF value. Valid entries are 0 to 55. The default is 30.

Defaults

30

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command is applicable only to VoIP dial peers.

Use the **icpif** command to specify the maximum acceptable impairment factor for the voice calls sent by the selected dial peer.

Examples

The following example disables the **icpif** command:

```
dial-peer voice 10 voip
 icpif 0
```

ignore

To specify the E&M or E&M MEL CAS voice port on the Cisco MC3810 to ignore specific receive bits, use the **ignore** command in voice-port configuration mode. Use the **no** form of this command to restore the default value.

ignore {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}

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

Syntax Description

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

Defaults

The default is mode-dependent:

E&M:

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

E&M MEL CAS:

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

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies only to digital voice ports on the Cisco MC3810.

Use this command with the **define** command.

Examples

To configure voice-port 1/1 to ignore receive bits b, c, and d, enter the following commands:

```
voice-port 1/1
 ignore rx-b-bit
 ignore rx-c-bit
 ignore rx-d-bit
```

Related Commands

Command	Description
condition	Manipulates the signalling format bit-pattern for all voice signalling types on the Cisco MC3810 multiservice concentrator.
define	Defines the send and receive bits for E&M and E&M MEL CAS voice signalling on the Cisco MC3810 multiservice concentrator.

image encoding

To select a specific encoding method for fax images associated with an MMoIP dial peer, use the **image encoding** command in dial-peer configuration mode. Use the **no** form of this command to restore the default value.

image encoding{**mh** | **mr** | **mmr** | **passthrough**}

no image encoding{**mh** | **mr** | **mmr** | **passthrough**}

Syntax Description

mh	Specifies Modified Huffman image encoding. This is the IETF standard.
mr	Specifies Modified Read image encoding.
mmr	Specifies Modified Modified Read image encoding.
passthrough	Specifies that the image will not be modified by an encoding method.

Defaults

Passthrough

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Use the **image encoding** command to specify an encoding method for e-mail fax TIFF images for a specific MMoIP dial peer. This command applies primarily to the on-ramp MMoIP dial peer. Although you can optionally create an off-ramp dial peer and configure a particular image encoding value for that off-ramp call leg, Store and Forward Fax ignores the off-ramp MMoIP setting and sends the file using Modified Huffman encoding.

There are four available encoding methods:

- Modified Huffman (MH)—One-dimensional data compression scheme that compresses data in only one direction (horizontal). Modified Huffman compression does not allow the transmission of redundant data. This encoding method produces the largest image file size.
- Modified Read (MR)—Two-dimensional data compression scheme (used by fax devices) that handles the data compression of the vertical line and that concentrates on the space between lines and within given characters.
- Modified Modified Read (MMR)—Data compression scheme used by newer Group 3 fax devices. This encoding method produces the smallest possible image file size and is slightly more efficient than Modified Read.
- Passthrough—No encoding method will be applied to the image—meaning that the image will be encoded by whatever encoding method is used by the fax device.

The IETF standard for sending fax TIFF images is Modified Huffman encoding with fine or standard resolution. RFC 2301 requires compliant receivers support TIFF images with MH encoding and fine or standard resolution. If a receiver supports features beyond this minimal requirement, you might want to configure the Cisco AS5300 to send enhanced-quality documents to that receiver.

The primary reason to use a different encoding scheme from MH is to save network bandwidth. MH ensures interoperability with all Internet fax devices but it is the least efficient of the encoding schemes for sending fax TIFF images. For most images, MR is more efficient than MH and MMR is more efficient than MR. If you know the recipient is capable of receiving more efficient encodings than just MH, Store and Forward Fax allows you to send the most efficient encoding the recipient can process. For end-to-end closed networks, you can choose any encoding scheme because the off-ramp gateway can process MH, MR, and MMR.

Another factor to take into account is the viewing software. Many viewing applications (for example, those that come with Windows 95 or Windows NT) are able to display MH, MR, and MMR. Therefore you should decide, based on the viewing application and the available bandwidth, which encoding scheme is right for your network.

This command applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

The following example selects Modified Modified Read as the encoding method for fax TIFF images sent by MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 image encoding mmr
```

Related Commands

Command	Description
image-resolution	Specifies a particular fax image resolution for a specific MMoIP dial peer.

image resolution

To specify a particular fax image resolution for a specific MMoIP dial peer, use the **image resolution** command in dial-peer configuration mode. Use the **no** form of this command to restore the default value.

image resolution {**fine** | **standard** | **superfine** | **passthrough**}

no connection {**fine** | **standard** | **superfine** | **passthrough**}

Syntax Description

fine	Configures the fax TIFF image resolution to be 204-by-196 pixels per inch.
standard	Configures the fax TIFF image resolution to be 204-by-98 pixels per inch.
super-fine	Configures the fax TIFF image resolution to be 204-by-391 pixels per inch.
passthrough	Indicates that the resolution of the fax TIFF image will not be altered.

Defaults

Passthrough

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Use the **image resolution** command to specify a specific resolution (in pixels per inch) for e-mail fax TIFF images sent by the specified MMoIP dial peer. This command applies primarily to the on-ramp MMoIP dial peer. Although you can optionally create an off-ramp dial peer and configure a particular image resolution value for that off-ramp call leg, Store and Forward Fax ignores the off-ramp MMoIP setting and sends the file using fine resolution.

This command enables you to increase or decrease the resolution of a fax TIFF image, thereby changing not only the resolution but the size of the fax TIFF file. The IETF standard for sending fax TIFF images is Modified Huffman encoding with fine or standard resolution. The primary reason to configure a different resolution is to save network bandwidth.

This command applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

The following example selects the resolution fine (meaning 204-by-196 pixels per inch) for e-mail fax TIFF images associated with MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 image encoding mh
 image resolution fine
```

Related Commands

Command	Description
image encoding	Selects a specific encoding method for fax images associated with an MMoIP dial peer.

impedance

To specify the terminating impedance of a voice port interface, use the **impedance** command in voice-port configuration mode. Use the **no** form of this command to restore the default value.

impedance {**600c** | **600r** | **900c** | **complex1** | **complex2**}

no impedance {**600c** | **600r** | **900c** | **complex1** | **complex2**}

Syntax Description

600c	Specifies 600 ohms complex.
600r	Specifies 600 ohms real.
900c	Specifies 900 ohms complex.
complex1	Specifies Complex 1.
complex2	Specifies Complex 2.

Defaults

600r

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **impedance** command to specify the terminating impedance of an FXO voice-port interface. The impedance value selected needs to match the specifications from the specific telephony system to which it is connected. Different countries often have different standards for impedance. CO switches in the United States are predominantly 600r. PBXs in the United States are normally either 600r or 900c.

If the impedance is set incorrectly (if there is an impedance mismatch), there will be a significant amount of echo generated (which could be masked if the **echo-cancel** command has been enabled). In addition, gains might not work correctly if there is an impedance mismatch.

Configuring the impedance on a voice port will change the impedance on both voice ports of a VPM card. This voice port must be shut down and then opened for the new value to take effect.

This command is applicable to FXS, FXO, and E&M voice ports on both the Cisco 3600 series and the Cisco MC3810.

Examples

The following example configures an FXO voice port on the Cisco 3600 series for a terminating impedance of 600 ohms (real):

```
voice-port 1/0/0
 impedance 600r
```

The following example configures an E&M voice port on the Cisco MC3810 for a terminating impedance of 900 ohms (complex):

```
voice-port 1/1
 impedance 900c
```

incoming called-number

To specify an incoming called number of an MMoIP or POTS dial peer, use the **incoming called-number** command in dial-peer configuration mode. Use the **no** form of this command to reset the default value for this command.

incoming called-number *string*

no incoming called-number *string*

Syntax Description

string	Specifies the incoming called telephone number. Valid entries are any series of digits that specify the E.164 telephone number.

Defaults

No incoming called number is defined.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.

Usage Guidelines

When a Cisco device (such as a Cisco AS5300 or Cisco AS5800) is handling both modem and voice calls, it needs to be able to identify the service type of the call—meaning whether 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.

If you do not use the **incoming called-number** command, 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.

By default, there is no called number associated with the dial peer, which means that incoming calls will be associated with dial peers based on matching calling number with answer address, call number with destination pattern, or calling interface with configured interface.

Use the **incoming-called number** command to define the destination telephone number for a particular dial peer. For the on-ramp POTS dial peer, this telephone number is the DNIS number of the incoming fax call. For the off-ramp MMoIP dial peer, this telephone number is the destination fax machine telephone number.

This command applies to both VoIP and POTS dial peers and applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

The following example configures calls coming in to the server with a called number of 3799262 as being voice calls:

```
dial peer voice 10 pots
 incoming called-number 3799262
```

The following example configures the number (310)555-9261 as the incoming called number for MMoIP dial peer 10:

```
dial-peer voice 10 mmoip
 incoming 3105559261
```

information-type

To select a particular information type for either an MMoIP or a POTS dial peer, use the **information-type** command in dial-peer configuration mode. Use the **no** form of this command to reset the default value for this command.

information type {**fax** | **voice**}

no information type {**fax** | **voice**}

Syntax Description

fax	Indicates that the information type has been set to store and forward fax.
voice	Indicates that the information type has been set to voice.

Defaults

Voice

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.

Usage Guidelines

This command applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

The following example sets the information type for MMoIP dial peer 10 to fax:

```
dial-peer voice 10 mmoip
 information-type fax
```

input gain

To configure a specific input gain value, use the **input gain** command in voice-port configuration mode. Use the **no** form of this command to disable the selected amount of inserted gain.

input gain *value*

no input gain *value*

Syntax Description

value	Specifies, in decibels, the amount of gain to be inserted at the receiver side of the interface. Acceptable value is any integer from –6 to 14.

Defaults

0

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

A system-wide loss plan must be implemented using both **input gain** and **output attenuation** commands. Other equipment (including PBXs) in the system must be taken into account when creating a loss plan. This default value for this command assumes that a standard transmission loss plan is in effect, meaning that normally, there must be –6 dB attenuation between phones. Connections are implemented to provide –6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0.

Please note that you cannot increase the gain of a signal going out into the PSTN, but you can decrease it. Therefore, if the voice level is too high, you can decrease the volume by either decreasing the input gain value or by increasing the output attenuation.

You can increase the gain of a signal coming in to the router. If the voice level is too low, you can increase the input gain.

Examples

The following example configures a 3-decibel gain to be inserted at the receiver side of the interface in the Cisco 3600 series:

```
port 1/0/0
 input gain 3
```

The following example configures a 3-decibel gain to be inserted at the receiver side of the interface in the Cisco MC3810:

```
port 1/1
 input gain 3
```

Related Commands

Command	Description
output attenuation	Configures a specific output attenuation value for a voice port.

ip precedence (dial-peer)

To set IP Precedence (priority) for packets sent by the dial peer, use the **ip precedence** command in dial-peer configuration mode. To restore the default value for this command, use the **no** form of this command.

ip precedence *number*

no ip precedence *number*

Syntax Description

number	Integer specifying the IP Precedence value. Valid entries are from 0 to 7. A value of 0 means that no precedence (priority) has been set.

Defaults

The default value for this command is 0.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use the **ip precedence** (dial-peer) command to configure the value set in the IP Precedence field when voice data packets are sent over the IP network. This command should be used if the IP link utilization is high and the quality of service for voice packets needs to have a higher priority than other IP packets. The **ip precedence** (dial-peer) command should also be used if RSVP is not enabled and the user would like to give voice packets a higher priority over other IP data traffic.

This command applies to VoIP peers.

Examples

The following example sets the IP Precedence to 5:

```
dial-peer voice 10 voip
 ip precedence 5
```

ip udp checksum

To calculate the UDP checksum for voice packets sent by the dial peer, use the **ip udp checksum** command in dial-peer configuration mode. Use the **no** form of this command to disable this feature.

ip udp checksum

no ip udp checksum

Syntax Description

This command has no arguments or keywords.

Defaults

disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **ip udp checksum** command to enable UDP checksum calculation for each of the outbound voice packets. This command is disabled by default to speed up the transmission of the voice packets. If you suspect that the connection has a high error rate, you should enable **ip udp checksum** to prevent corrupted voice packets forwarded to the digital signal processor (DSP).

This command applies to VoIP peers.

Examples

The following example calculates the UDP checksum for voice packets sent by this dial peer:

```
dial-peer voice 10 voip
 ip udp checksum
```

Related Commands

Command	Description
loop-detect	Enables loop detection for T1 for Voice over ATM, Voice over Frame Relay, and Voice over HDLC.

ivr autoload

To load files from a particular TFTP server (as indicated by a defined URL), use the **ivr autoload** command in global configuration mode. To disable this function, use the **no** form of this command.

ivr autoload url *location*

no ivr autoload url *location*

Syntax Description

url	Indicates that a URL is used to locate the index file that contains a list of all available audio files.
location	Specifies the URL of the index file. Example of index file on TFTP: `tftp://keyer/index` Example of index file on Flash: `flash:index`

Defaults

No URL is defined.

Command Modes

Global configuration

Command History

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

Usage Guidelines

The index file contains a list of audio files (URL) that can be downloaded from the TFTP server. Use the **ivr autoload** command to download audio files from TFTP to memory. The command only starts up a background process. The background process (loader) does the actual down-loading of the files.

The background process first reads the index file from either Flash or TFTP. It parses the files line by line looking for the URL. It ignores lines which starts with # as comment lines. Once it has a correct URL, it tries to read that .au file into memory and creates a media object. If there are any errors during the reading of the file, it retries the configured number of times. If *mode* is set to 'verbose' the loader logs the transaction to console. Once parsing has reached the end of the index file, the background process exits out of memory.

Perform the following checks before initiating the background process. If one of the checks fail, it indicates the background process is not started, and instead you will see an error response to the command.

- Check if any prompt is being actively used (IVR is actively playing some prompts). If there are active prompts, the command fails with following error (.au files are also referred to as prompts):

```
command is not allowed when prompts are active
```

- Check if there is already a back-ground process in progress. If there a process, command fails with following error:

```
previous autoload command is still in progress
```

- Check if there is already a earlier **ivr autoload** command. If there is already an **ivr autoload** command configured, the user sees the following response when the command is issued:

  ```
  previous command is being replaced
  ```

- When the **no ivr autoload** command is issued, if there was already an **ivr autoload** command in progress, it will be aborted.

The audio files (prompts) loaded using the **ivr autoload** command are not dynamically swapped out of memory. They are considered as autoloaded prompts as opposed to "dynamic" prompts. (See the **ivr prompt memory** command for details on dynamic prompts.)

Examples

The following example loads audio files from the TFTP server (located at //jurai/mgindi/tclware/index4):

```
ivr autoload url tftp://jurai/mgindi/tclware/index4
```

The index file for this example index4 is shown as:

```
Router# more index4
tftp://jurai/mgindi/tclware/au/en/en_one.au
#tftp://jurai/mgindi/tclware/au/ch/ch_one.au
tftp://jurai/mgindi/tclware/au/ch/ch_one.au
```

Related Commands

Command	Description
ivr prompt memory	Configures the maximum amount of memory you wish to allow the dynamic audio files (prompts) to occupy in memory.

ivr autoload retry

To specify the number of times the system will try to load audio files from TFTP to memory when there is an error, use the **ivr autoload retry** command in global configuration mode. To disable this function, use the **no** form of this command.

ivr autoload retry *number*

no ivr autoload retry *number*

Syntax Description

number	Indicates a number from 1 to 5. The default value is 3.

Defaults

ivr autoload retry 3

Command Modes

Global configuration

Command History

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

Examples

The following example configures the system to try three times to load audio files:

```
ivr autoload retry 3
```

Related Commands

Command	Description
ivr prompt memory	Configures the maximum amount of memory you wish to allow the dynamic audio files (prompts) to occupy in memory.

ivr autoload mode

To load files from TFTP to memory using either silent or verbose mode, use the **ivr autoload mode** command in global configuration mode. To disable this function, use the **no** form of this command.

ivr autoload mode {**verbose url** *location* [**retry** *number*]} | {**silent url** *location* [**retry** *number*]}

no ivr autoload mode {**verbose url** *location* [**retry** *number*]} | {**silent url** *location* [**retry** *number*]}

Syntax Description

verbose	Displays the file transfer activity to the console. This mode is recommended for use while debugging.
silent	Performs the file transfer in silent mode, meaning that no file transfer activity is displayed to the console.
retry	(Optional) Specifies the number of times the system will try to transfer a file when there are errors. This parameter applies to each file transfer.
number	(Optional) Indicates a number from 1 to 5. The default value is 3.
url	Indicates that a URL is used to locate the index file that contains a list of all available audio files.
location	Specifies the URL of the index file. Example of index file on TFTP: `tftp://keyer/index` Example of index file on Flash: `flash:index`

Defaults

ivr autoload silent

Command Modes

Global configuration

Command History

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

Usage Guidelines

The index file contains a list of audio files (URL) that can be downloaded from the TFTP server. Use the **ivr autoload** command to download audio files from TFTP to memory. The command only starts up a background process. The background process (loader) does the actual downloading of the files.

The background process first reads the index file from either Flash or TFTP. It parses the files line by line looking for the URL. It ignores lines which starts with # as comment lines. Once it has a correct URL, it tries to read that .au file into memory and creates a media object. If there are any errors during the reading of the file, it retries the configured number of times. If mode is set to **verbose**, the loader logs the transaction to console. Once parsing has reached the end of the index file, the background process exits out of memory.

Perform the following checks before initiating the background process. If one of the checks fail, it indicates the background process is not started, and instead you will see an error response to the command.

- Check if any prompt is being actively used (IVR is actively playing some prompts). If there are active prompts, the command fails with following error (.au files are also referred to as prompts):

  ```
  command is not allowed when prompts are active
  ```

- Check if there is already a back-ground process in progress. If there is a process, command fails with following error:

  ```
  previous autoload command is still in progress
  ```

- Check if there is already a earlier **ivr autoload** command. If there is already an **ivr autoload** command configured, the user sees the following response when the command is issued:

  ```
  previous command is being replaced
  ```

- When the **no ivr autoload** command is issued, if there was already an **ivr autoload** command in progress, it will be aborted.

The audio files (prompts) loaded using the **ivr autoload** command are not dynamically swapped out of memory. They are considered as autoloaded prompts as opposed to "dynamic" prompts. (See the **ivr prompt memory** command for details on dynamic prompts.)

Examples

The following example configures verbose mode:

```
ivr autoload mode verbose retry 3 url tftp://jurai/mgindi/tclware/index4
```

The index file for this example index4 is shown as:

```
Router# more index4
tftp://jurai/mgindi/tclware/au/en/en_one.au
#tftp://jurai/mgindi/tclware/au/ch/ch_one.au
tftp://jurai/mgindi/tclware/au/ch/ch_one.au
```

Related Commands

Command	Description
ivr prompt memory	Configures the maximum amount of memory you wish to allow the dynamic audio files (prompts) to occupy in memory.

ivr prompt memory

To configure the maximum amount of memory for the dynamic audio files (prompts) to occupy in memory, use the **ivr prompt memory** command in global configuration mode. Use the **no** form of this command to disable the maximum memory size.

ivr prompt memory *size* **files** *number*

no ivr prompt memory *size* **files** *number*

Syntax Description

size	Specifies the maximum memory to be used by the free dynamic prompts in kilobytes. Valid entries are from 128 to 16384.
files *number*	Specifies the number of files that can stay in memory. Valid entries for the number argument is 50 to 1000.

Command Modes

Global configuration

Command History

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

Usage Guidelines

When both the *number* and *size* parameters are specified, the minimum memory out of the two will be used for memory calculations.

All the prompts which are not autoloaded or fixed are considered as dynamic. Dynamic prompts are loaded in to memory from TFTP or Flash, as and when they are needed. When they are actively used for playing prompts they are considered to be in 'active' state. However, once the prompt playing is complete, these prompts are no more active and are considered to be in 'free' state.

The free prompts either stay in memory or removed out of memory depending on the availability of space in memory for these free prompts. The **prompt-mem** command essentially specifies a maximum memory to be used for these free prompts.

The free prompts are saved in the memory and are queued in a waitQ. When the waitQ is full (either because the totally memory occupied by the free prompts exceeds the max. configured value or the number of files in the waitQ exceeds max. configured), oldest free prompts are removed out of memory.

Examples

The following example shows how to use the **ivr prompt memory** command:

```
ivr prompt memory 2048 files 500
```

Related Commands

Command	Description
ivr autoload	Loads files from TFTP to memory.
show call prompt-mem-usage	Displays the memory site use by prompts.

loopback (T1 controller)

To set the loopback method for testing the T1 interface, enter the **loopback** command in controller configuration mode. Use the **no** form of this command to restore the default value.

loopback {**diagnostic** | **local** {**payload** | **line**} | **remote** {**iboc** | **esf** {**payload** | **line**}}

no loopback

Syntax Description

diagnostic	Loops the outgoing transmit signal back to the receive signal
line	Places the interface into external loopback mode at the line.
local	Places the interface into local loopback mode.
payload	Places the interface into external loopback mode at the payload level.
remote	Keeps the local end of the connection in remote loopback mode.
iboc	Sends an in-band bit-oriented code to the far end to cause it to go into line loopback.
esf	Specifies extended super frame as the T1 or E1 frame type.

Defaults

No loopback is configured.

Command Modes

Controller configuration

Command History

Release	Modification
11.3 MA	This command was introduced as a controller configuration command for the Cisco MC3810.
12.0(5)T and 12.0(5)XK	The command was introduced as an ATM interface configuration command for the Cisco 2600 and 3600 series.
12.0(5)XE	The command was introduced as an ATM interface configuration command for the Cisco 7200 and 7500 series.
12.0(5)XK and 12.0(7)T	The command was introduced as a controller configuration command for the Cisco 2600 and 3600 series.

Usage Guidelines

You can use a loopback test on lines to detect and distinguish equipment malfunctions caused either by line and channel service unit/digital service unit (CSU/DSU) or by the interface. If correct data transmission is not possible when an interface is in loopback mode, the interface is the source of the problem.

Examples

The following example shows how to set the diagnostic loopback method on controller T1 0/0:

```
Router(config)# controller t1 0/0
loopback diagnostic
```

loop-detect

To enable loop detection for T1, use the **loop-detect** command in controller configuration mode. Use the **no** form of the command to cancel the loop detect operation.

loop-detect

no loop-detect

Syntax Description

This command has no arguments or keywords.

Defaults

Loop detection is disabled.

Command Modes

Controller configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Examples

The following example configures loop detection for controller T1 0:

```
controller t1 0
 loop-detect
```

Related Commands

Command	Description
loopback (interface)	Diagnoses equipment malfunctions between an interface and a device.

loss-plan

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

loss-plan {**plan1** | **plan2** | **plan5** | **plan6** | **plan7**}

no loss-plan

Syntax Description

plan1	FXO: A-D gain = 0 db, D-A gain = 0 db FXS: A-D gain = -3 db, D-A gain = –3 db
plan2	FXO: A-D gain = 3 db, D-A gain = 0 db FXS: A-D gain = 0 db, D-A gain =–3 db
plan5	FXO: Not applicable FXS: A-D gain = –3 db, D-A gain = –10 db
plan6	FXO: Not applicable FXS: A-D gain = 0 db, D-A gain = –7 db
plan7	FXO: A-D gain = 7 db, D-A gain = 0 db FXS: A-D gain = 0 db, D-A gain = –6 db

Defaults

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

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

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.

Usage Guidelines

This command applies to the Cisco MC3810 only.

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

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

Examples

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

```
voice-port 1/6
 loss-plan plan3
```

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

```
voice-port 1/1
 loss-plan plan6
```

Related Commands

Command	Description
impedance	Specifies the terminating impedance (amount of wire resistance and reactivity to current) of a voice port interface. The setting must match the physical wiring.
input gain	Configures a specific input gain value for a voice port.
output attenuation	Configures a specific output attenuation value for a voice port.

lrq forward-queries

To enable a gatekeeper to forward Location Requests (LRQs) that contain E.164 addresses that match zone prefixes controlled by remote gatekeepers, use the **lrq forward-queries** command in gatekeeper configuration mode. To disable this function, use the **no** form of this command.

lrq forward-queries

no lrq forward-queries

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Gatekeeper configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

LRQ forwarding is dependent on a Cisco non-standard field that first appeared in Cisco IOS Release 12.0(3)T. This means that any LRQ received from a non-Cisco gatekeeper or any gatekeeper running a Cisco IOS software image prior to Cisco IOS Release 12.0(3)T will not be forwarded.

The routing of E.164-addressed calls is dependent on the configuration of zone prefix tables (for example, area code definitions) on each gatekeeper. Each gatekeeper is configured with a list of prefixes controlled by itself and by other remote gatekeepers. Calls are routed to the zone that manages the matching prefix. Thus, in the absence of a directory service for such prefix tables, you, the network administrator, may have to define extensive lists of prefixes on all the gatekeepers in your administrative domain.

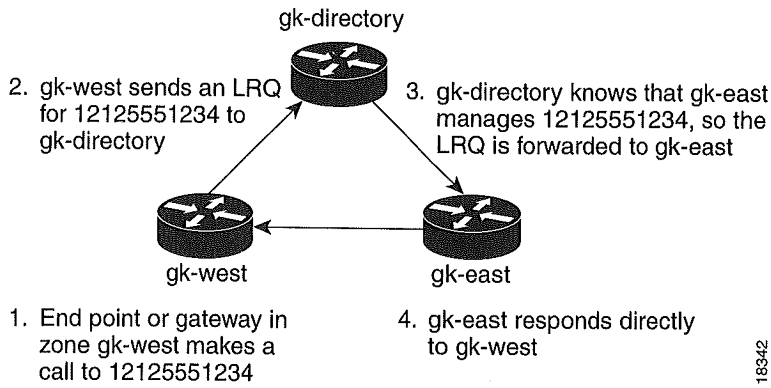
To simplify this task, you can select one of your gatekeepers as the "directory" gatekeeper and configure that gatekeeper with the complete list of prefixes and the **lrq forward-queries** command. You can then simply configure all the other gatekeepers with their own prefixes and the wildcard prefix "*" for your directory gatekeeper.

This command only affects the forwarding of LRQs for E.164 addresses. LRQs for H.323-ID addresses are never forwarded.

Examples

The following example shows how this command is used to simplify configuration by selecting one gatekeeper as the directory gatekeeper. Refer to Figure 3.

Figure 3 Example Scenario with Directory Gatekeeper and Two Remote Gatekeepers



Configuration on gk-directory

On the directory gatekeeper called gk-directory, identify all the prefixes for all the gatekeepers in your administrative domain:

```
gk-directory(config-gk)# zone local gk-directory cisco.com
gk-directory(config-gk)# zone remote gk-west cisco.com 172.0.1.1
gk-directory(config-gk)# zone remote gk-east cisco.com 172.0.2.1

gk-directory(config-gk)# zone prefix gk-west 1408.......
gk-directory(config-gk)# zone prefix gk-west 1415.......
gk-directory(config-gk)# zone prefix gk-west 1213.......
gk-directory(config-gk)# zone prefix gk-west 1650.......

gk-directory(config-gk)# zone prefix gk-east 1212.......
gk-directory(config-gk)# zone prefix gk-east 1617.......

gk-directory(config-gk)# lrq forward-queries
```

Configuration on gk-west

On the gatekeeper called gk-west, configure all the locally managed prefixes for that gatekeeper:

```
gk-west(config-gk)# zone local gk-west cisco.com
gk-west(config-gk)# zone remote gk-directory cisco.com 172.1.2.3

gk-west(config-gk)# zone prefix gk-west 1408.......
gk-west(config-gk)# zone prefix gk-west 1415.......
gk-west(config-gk)# zone prefix gk-west 1213.......
gk-west(config-gk)# zone prefix gk-west 1650.......
gk-west(config-gk)# zone prefix gk-directory *
```

Configuration on gk-east

On the gatekeeper called gk-east, configure all the locally managed prefixes for that gatekeeper:

```
gk-east(config-gk)# zone local gk-east cisco.com
gk-east(config-gk)# zone remote gk-directory cisco.com 172.1.2.3

gk-east(config-gk)# zone prefix gk-east 1212.......
gk-east(config-gk)# zone prefix gk-east 1617.......
gk-east(config-gk)# zone prefix gk-directory *
```

Now when an endpoint or gateway in zone gk-west makes a call to 12125551234, gk-west will send an LRQ for that E.164 address to gk-directory, which forwards the LRQ to gk-east. Gatekeeper gk-east responds directly to gk-west.

Related Commands

Command	Description
lrq reject-unknown-prefix	Enables the gatekeeper to reject all Location Requests (LRQs) for zone prefixes that are not configured.

lrq reject-unknown-prefix

To enable the gatekeeper to reject all Location Requests (LRQs) for zone prefixes that are not configured, use the **lrq reject-unknown-prefix** command in gatekeeper configuration mode. To reenable the gatekeeper to accept and process all incoming LRQs, use the **no** form of this command.

lrq reject-unknown-prefix

no lrq reject-unknown-prefix

Syntax Description

This command has no arguments or keywords.

Defaults

The gatekeeper accepts and processes all incoming LRQs.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Usage Guidelines

Use the **lrq reject-unknown-prefix** command to configure the gatekeeper to reject any incoming LRQs for a destination E.164 address that does not match any of the configured zone prefixes.

Whether or not you enable the **lrq reject-unknown-prefix** command, the following is true when the E.164 address matches a zone prefix:

- If the matching zone prefix is local (that is, controlled by this gatekeeper), the LRQ is serviced.
- If the matching zone prefix is remote (that is, controlled by some other gatekeeper), the LRQ is rejected.

If you do not enable the **lrq reject-unknown-prefix** command, and the target address does not match any known local or remote prefix, the default behavior is to attempt to service the call using one of the local zones. If this default behavior is not suitable for your site, configure the **lrq reject-unknown-prefix** command on your router to force the gatekeeper to reject such requests.

Examples

Consider the following gatekeeper configuration:

```
zone local gk408 cisco.com
zone local gk415 cisco.com
zone prefix gk408 1408.......
zone prefix gk415 1415.......
lrq reject-unknown-prefix
```

In this example configuration, the gatekeeper is configured to manage two zones. One zone contains gateways with interfaces in the 408 area code, and the second zone contains gateways in the 415 area code. Then using the **zone prefix** command, the gatekeeper is configured with the appropriate prefixes so that calls to those area codes hop off in the optimal zone.

Now say some other zone has been erroneously configured to route calls to the 212 area code to this gatekeeper. When the LRQ for a number in the 212 area code arrives at this gatekeeper, the gatekeeper fails to match the area code, and the LRQ is rejected.

If this was your only site that had any gateways in it, and you wanted your other sites to route all calls requiring gateways to this gatekeeper, then you can undo the **lrq reject-unknown-prefix command** by simply using the **no lrq reject-unknown-prefix**. Now when the gatekeeper receives an LRQ for the address 12125551234, it will attempt to find an appropriate gateway in either one of the zones gk408 or gk415 to service the call.

Related Commands

Command	Description
lrq forward-queries	Enables a gatekeeper to forward Location Requests (LRQs) that contain E.164 addresses that match zone prefixes controlled by remote gatekeepers.

max-conn

To specify a maximum number of allowed connections for a particular dial peer, use the **max-conn** command in dial-peer configuration mode. Use the **no** form of this command to set an unlimited number of connections for this dial peer.

max-conn *number*

no max-conn *number*

Syntax Description

number	Specifies the maximum number of connections for this dial peer. Valid values for this field are 1 to 2147483647.

Defaults

The **no** form of this command is the default, meaning unlimited number of connections.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.

Usage Guidelines

This command applies to both VoIP and POTS dial peers.

Use the **max-conn** command to define the maximum number of connections used simultaneously on the Cisco AS5300 to send fax-mail.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example configures the maximum number of connections for VoIP dial peer 10 as 5:

```
dial-peer voice 10 voip
 max-conn 5
```

Related Commands

Command	Description
mta receive maximum-recipients	Specifies the maximum recipients for all SMTP connections.

max-connection

To set the maximum number of simultaneous connections to be used for communication with a settlement provider, use the **max-connection** command in settlement configuration mode. Use the **no** form of this command to reset to the default value of this command.

max-connection *num*

no max-connection *num*

Syntax Description

num	Specifies the maximum number of HTTP connections to a settlement provider.

Defaults

The default is 20 maximum connections.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Examples

The following command sets the maximum number of simultaneous connections to be 10:

```
settlement 0
 max-connections 10
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of connection attempts to the provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement configuration mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.
url	Configures the ISP address.

mdn

To request that a message disposition notice (MDN) be generated when the message is processed ("opened"), use the **mdn** command in dial-peer configuration mode. Use the **no** form of this command to restore the default value.

mdn

no mdn

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Message disposition notification is an e-mail message that is generated and sent to the sender when the message is opened by the receiver. Use the **mdn** command to request that an e-mail response message be sent to the sender when the e-mail containing the fax TIFF image has been opened.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example requests that a message disposition notice be generated by the recipient:

```
dial-peer voice 10 mmoip
 mdn
```

Related Commands

Command	Description
mta receive generate-mdn	Specifies that the off-ramp gateway process a response MDN from an SMTP server.
mta send-return receipt-to	Specifies the address where MDNs will be sent.

mmoip aaa global-password

To define a password to be used with CiscoSecure for Windows NT when using Store and Forward Fax, use the **mmoip aaa global-password** command in global configuration mode. Use the **no** form of this command to restore the default value.

mmoip aaa global-password *password*

no mmoip aaa global-password *password*

Syntax Description

password	Character string used to define the CiscoSecure for Windows NT password to be used with Store and Forward Fax. Maximum length is 64 alphanumeric characters.

Defaults

No password defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

CiscoSecure for Windows NT might require a separate password in order to complete authentication, no matter what security protocol you use. This command defines the password to be used with CiscoSecure for Windows NT. All records on the Windows NT server use this defined password.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example defines a password (abercrombie) when CiscoSecure for Windows NT is used with Store and Forward Fax:

```
configure terminal
 mmoip aaa global-password abercrombie
```

mmoip aaa method fax accounting

To define the name of the method list to be used for AAA accounting with Store and Forward Fax, use the **mmoip aaa method fax accounting** command in global configuration mode. Use the **no** form of this command to restore the default value.

mmoip aaa method fax accounting *method-list-name*

no mmoip aaa method fax accounting *method-list-name*

Syntax Description

method-list-name	Character string used to name a list of accounting methods to be used with Store and Forward Fax.

Defaults

No AAA accounting method list defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command defines the name of the AAA accounting method list to be used with Store and Forward Fax. The method list itself, which defines the type of accounting services provided for Store and Forward Fax, is defined using the **aaa accounting** global configuration command. Unlike standard AAA (where each defined method list can be applied to specific interfaces and lines), the AAA accounting method lists used in Store and Forward Fax are applied globally on the Cisco AS5300.

After the accounting method lists have been defined, they are enabled by using the **mmoip aaa receive-accounting enable** command.

This command applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

The following example defines a AAA accounting method list (called sherman) to be used with Store and Forward Fax:

```
configure terminal
 aaa new-model
 mmoip aaa method fax accounting sherman
```

Related Commands

Command	Description
mmoip aaa receive-accounting enable	Enables on-ramp Store and Forward Fax AAA accounting services.

mmoip aaa method fax authentication

To define the name of the method list to be used for AAA authentication with Store and Forward Fax, use the **mmoip aaa method fax authentication** command in global configuration mode. Use the **no** form of this command to restore the default value.

mmoip aaa method fax authentication *method-list-name*

no mmoip aaa method fax authentication *method-list-name*

Syntax Description

method-list-name	Character string used to name a list of authentication methods to be used with Store and Forward Fax.

Defaults

No AAA authentication method list defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command defines the name of the AAA authentication method list to be used with Store and Forward Fax. The method list itself, which defines the type of authentication services provided for Store and Forward Fax, is defined using the **aaa authentication** global configuration command. Unlike standard AAA (where each defined method list can be applied to specific interfaces and lines), AAA authentication method lists used with Store and Forward Fax are applied globally on the Cisco AS5300.

After the authentication method lists have been defined, they are enabled by using the **mmoip aaa receive-authentication enable** command.

This command applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

The following example defines a AAA authentication method list (called peabody) to be used with Store and Forward Fax:

```
configure terminal
 aaa new-model
 mmoip aaa method fax authentication peabody
```

Related Commands

Command	Description
mmoip aaa receive-authentication enable	Enables on-ramp Store and Forward Fax AAA authentication services.

mmoip aaa receive-accounting enable

To enable on-ramp AAA accounting services, use the **mmoip aaa receive-accounting enable** command in global configuration mode. Use the **no** form of this command to restore the default value.

mmoip aaa receive-accounting enable

no mmoip aaa receive-accounting enable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command enables AAA accounting services if a AAA accounting method list has been defined using both the **aaa accounting** command and the **mmoip aaa method fax accounting** command.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example enables a AAA accounting method list (called sherman) to be used with inbound Store and Forward Fax. In this example, Store and Forward Fax is being configured to track start and stop connection accounting records.

```
configure terminal
 aaa new-model
 mmoip aaa method fax accounting sherman
 aaa accounting connection sherman stop-only radius
 mmoip aaa receive-accounting enable
```

Related Commands

Command	Description
mmoip aaa method fax accounting	Defines the name of the method list to be used for AAA accounting with Store and Forward Fax.

mmoip aaa receive-authentication enable

To enable on-ramp AAA authentication services, use the **mmoip aaa receive-authentication enable** command in global configuration mode. Use the **no** form of this command to restore the default value.

mmoip aaa receive-authentication enable

no mmoip aaa receive-authentication enable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command enables AAA authentication services if an AAA authentication method list has been defined using both the **aaa authentication** command and the **mmoip aaa method fax authentication** command.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example enables a AAA authentication method list (called peabody) to be used with inbound Store and Forward Fax. In this example, RADIUS (and if the RADIUS server fails, then local) authentication is being configured for Store and Forward Fax.

```
configure terminal
 aaa new-model
 mmoip aaa method fax authentication peabody
 aaa authentication login peabody radius local
 mmoip aaa receive-authentication enable
```

Related Commands

Command	Description
mmoip aaa method fax authentication	Defines the name of the method list to be used for AAA authentication with Store and Forward Fax.

mmoip aaa receive-id primary

To specify the primary location where AAA retrieves its account identification information for on-ramp faxing, use the **mmoip aaa receive-id primary** command in global configuration mode. Use the **no** form of this command to restore the default value, which means that account identification source is undefined.

mmoip aaa receive-id primary {**ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}

no mmoip aaa receive-id primary {**ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}

Syntax Description

ani	Indicates that AAA uses the calling party telephone number (automatic number identification or ANI) as the AAA account identifier.
dnis	Indicates that AAA uses the called party telephone number (dialed number identification service or DNIS) as the AAA account identifier.
gateway	Indicates that AAA uses the router-specific name derived from the host name and domain name as the AAA account identifier, displayed in the following format: *router-name.domain-name*.
redialer-id	Indicates that AAA uses the account string returned by the external redialer device as the AAA account identifier. In this case, the redialer ID is either the redialer serial number or the redialer account number.
redialer-dnis	Indicates that AAA uses the called party telephone number (dialed number identification service or DNIS) as the AAA account identifier captured by the redialer if a redialer device is present.

Defaults

No account identification source is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Normally, when AAA is being used for simple user authentication, AAA uses the username information defined in the user profile for authentication. With Store and Forward Fax, you can specify that the ANI, DNIS, gateway ID, redialer ID, or redialer DNIS be used to identify the user for authentication. This command defines what AAA uses for the primary identifier for inbound or on-ramp user authentication with Store and Forward Fax.

Store and Forward Fax allows you to define either a primary or a secondary identifier. (You configure the secondary identifier using the **mmoip aaa receive-id secondary** command.)

AAA does not use these methods sequentially—meaning that if the primary identifier is defined and AAA cannot authenticate the primary identifier information, it will not use the secondary identifier for authentication. Authentication simply fails.

Defining only the secondary identifier enables you to service two different scenarios simultaneously—for example, if you are offering fax services to two different companies, one of which uses redialers and the other of which does not. In this case, configure the **mmoip aaa receive-id primary** command to use the redialer DNIS, and configure the **mmoip aaa receive id secondary** command to use ANI. With this configuration, when a user dials in and the redialer-DNIS is not null, then the redialer DNIS is used as the authentication identifier. If a user dials in and the redialer DNIS is null, then ANI is used as the authentication identifier.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example defines the DNIS captured by the redialer as the AAA authentication identifier for Store and Forward Fax:

```
configure terminal
 aaa new-model
 mmoip aaa receive-id primary redialer-dnis
```

Related Commands

Command	Description
mmoip aaa receive-id secondary	Specifies the secondary location where AAA retrieves its account identification information for on-ramp faxing if the primary identifier has not been defined.

mmoip aaa receive-id secondary

To specify the secondary location where AAA retrieves its account identification information for on-ramp faxing if the primary identifier has not been defined, use the **mmoip aaa receive-id secondary** command in global configuration mode. Use the **no** form of this command to restore the default value, which means that account identification source is undefined.

mmoip aaa receive-id secondary {**ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}

no mmoip aaa receive-id secondary {**ani** | **dnis** | **gateway** | **redialer-id** | **redialer-dnis**}

Syntax Description

ani	Indicates that AAA uses the calling party telephone number (automatic number identification or ANI) as the AAA account identifier.
dnis	Indicates that AAA uses the called party telephone number (dialed number identification service or DNIS) as the AAA account identifier.
gateway	Indicates that AAA uses the router-specific name derived from the host name and domain name as the AAA account identifier, displayed in the following format: *router-name.domain-name*.
redialer-id	Indicates that AAA uses the account string returned by the external redialer device as the AAA account identifier. In this case, the redialer ID is either the redialer serial number or the redialer account number.
redialer-dnis	Indicates that AAA uses the called party telephone number (dialed number identification service or DNIS) as the AAA account identifier captured by the redialer if a redialer device is present.

Defaults

No account identification source is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Normally, when AAA is being used for simple user authentication, AAA uses the username information defined in the user profile for authentication. With Store and Forward Fax, you can specify that the ANI, DNIS, gateway ID, redialer DNIS, or redialer ID be used to identify the user for authentication. This command defines what AAA uses for the secondary identifier for inbound or on-ramp user authentication with Store and Forward Fax if the primary identifier has not been defined.

Store and Forward Fax allows you to define either a primary or a secondary identifier. (You configure the primary identifier using the **mmoip aaa receive-id primary** command.)

AAA does not use these methods sequentially—meaning that if the primary identifier is defined and AAA cannot match the primary identifier information, it will not use the secondary identifier for authentication. Authentication simply fails.

Defining only the secondary identifier enables you to service two different scenarios simultaneously—for example, if you are offering fax services to two different companies, one of which uses redialers and the other of which does not. In this case, configure the **mmoip aaa receive-id primary** command to use the redialer DNIS, and configure the **mmoip aaa receive id secondary** command to use ANI. With this configuration, when a user dials in and the redialer-DNIS is not null, then the redialer DNIS is used as the authentication identifier. If a user dials in and the redialer DNIS is null, then ANI is used as the authentication identifier.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example defines the DNIS captured by the redialer as the secondary AAA authentication identifier for Store and Forward Fax:

```
configure terminal
 aaa new-model
 mmoip aaa receive-id secondary redialer-dnis
```

Related Commands

Command	Description
mmoip aaa receive-id primary	Specifies the primary location where AAA retrieves its account identification information for on-ramp faxing.

mmoip aaa send-accounting enable

To enable off-ramp AAA accounting services, use the **mmoip aaa send-accounting enable** command in global configuration mode. Use the **no** form of this command to restore the default value.

mmoip aaa send-accounting enable

no mmoip aaa send-accounting enable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command enables AAA accounting services if a AAA accounting method list has been defined using both the **aaa accounting** command and the **mmoip aaa method fax accounting** command.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example enables a AAA accounting method list (called sherman) to be used with outbound Store and Forward Fax. In this example, Store and Forward Fax is being configured to track start and stop connection accounting records.

```
configure terminal
 aaa new-model
 mmoip aaa method fax accounting sherman
 aaa accounting connection sherman stop-only radius
 mmoip aaa send-accounting enable
```

Related Commands

Command	Description
mmoip aaa method fax accounting	Defines the name of the method list to be used for AAA accounting with Store and Forward Fax.

mmoip aaa send-authentication enable

To enable off-ramp AAA authentication services, use the **mmoip aaa send-authentication enable** command in global configuration mode. Use the **no** form of this command to restore the default value.

mmoip aaa send-authentication enable

no mmoip aaa send-authentication enable

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command enables AAA authentication services if a AAA authentication method list has been defined using both the **aaa authentication** command and the **mmoip aaa method fax authentication** command.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example enables a AAA authentication method list (called peabody) to be used with outbound Store and Forward Fax. In this example, RADIUS (and if the RADIUS server fails, then local) authentication is being configured for Store and Forward Fax.

```
configure terminal
 aaa new-model
 mmoip aaa method fax authentication peabody
 aaa authentication login peabody radius local
 mmoip aaa send-authentication enable
```

Related Commands

Command	Description
mmoip aaa method fax authentication	Defines the name of the method list to be used for AAA authentication with Store and Forward Fax.

mmoip aaa send-id primary

To specify the primary location where AAA retrieves its account identification information for off-ramp faxing, use the **mmoip aaa send-id primary** command in global configuration mode. Use the **no** form of this command to restore the default value, which means that account identification source is undefined.

mmoip aaa send-id primary {**account-id** | **envelope-from** | **envelope-to** | **gateway**}

no mmoip aaa send-id primary {**account-id** | **envelope-from** | **envelope-to** | **gateway**}

Syntax Description

account-id	Indicates that AAA uses the account username from the originating fax-mail system as the AAA account identifier. This means the off-ramp gateway uses the account identifier in the x-account ID field of the e-mail header. The benefit of using this attribute offers end-to-end authentication and accounting tracking.
envelope-from	Indicates that AAA uses the account username from the fax-mail header as the AAA account identifier.
envelope-to	Indicates that AAA uses the recipient derived from the fax-mail header as the AAA account identifier.
gateway	Indicates that AAA uses the router-specific name derived from the host name and domain name as the AAA account identifier, displayed in the following format: *router-name.domain-name*.

Defaults

No account identification source is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Normally, when AAA is being used for simple user authentication, AAA uses the username information defined in the user profile for authentication. With Store and Forward Fax, you can specify that the account ID, username, or recipient name from the e-mail header information be used to identify the user for authentication. This command defines what AAA uses for the primary identifier for outbound or off-ramp user authentication with Store and Forward Fax.

Store and Forward Fax allows you to define either a primary or a secondary identifier. (You configure the secondary identifier using the **mmoip aaa send-id secondary** command.) AAA extracts the authentication identifier information from the defined sources. If the field is blank (meaning undefined), AAA will use the secondary identifier source if configured. The secondary identifier is used only when the primary identifier is null. In this case, when AAA sees that the primary identifier is null, it will check to see if a secondary identifier has been defined and use that value for user authentication.

AAA does not use these methods sequentially—meaning that if the primary identifier is defined and AAA cannot authenticate the primary identifier information, it will not use the secondary identifier for authentication. Authentication simply fails.

When you enable authentication, the on-ramp gateway inserts whatever value you configure for the **mmoip aaa receive-id primary** command in the X-account-ID field of the e-mail header. This X-account ID field contains the value that is used for authentication and accounting by the on-ramp gateway. For example, if the **mmoip aaa receive-id primary** command is set to **gateway**, the on-ramp gateway name (for example, hostname.domain-name) is inserted in the X-account ID field of the e-mail header of the fax-mail message.

If you want to use this configured gateway value in the X-account ID field, you must configure the **mmoip aaa send-id primary** command with the **account-id** keyword. This particular keyword enables Store and Forward Fax to generate end-to-end authentication and accounting tracking records. If you do not enable authentication on the on-ramp gateway, the X-account-ID field is left blank.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example defines the recipient name as defined in the envelope-to field of the e-mail header to be used as the AAA authentication identifier for Store and Forward Fax:

```
configure terminal
 aaa new-model
 mmoip aaa send-id primary envelope-to
```

Related Commands

Command	Description
mmoip aaa send-id secondary	Specifies the secondary location where AAA retrieves its account identification information for off-ramp faxing.

mmoip aaa send-id secondary

To specify the secondary location where AAA retrieves its account identification information for off-ramp faxing, use the **mmoip aaa send-id secondary** command in global configuration mode. Use the **no** form of this command to restore the default value, which means that account identification source is undefined.

mmoip aaa send-id secondary {**account-id** | **envelope-from** | **envelope-to** | **gateway**}

no mmoip aaa send-id secondary {**account-id** | **envelope-from** | **envelope-to** | **gateway**}

Syntax Description

account-id	Indicates that AAA uses the account username from the originating fax-mail system as the AAA account identifier. This means the off-ramp gateway uses the account identifier in the x-account ID field of the e-mail header. The benefit of using this attribute offers end-to-end authentication and accounting tracking.
envelope-from	Indicates that AAA uses the account username from the fax-mail header as the AAA account identifier.
envelope-to	Indicates that AAA uses the recipient derived from the fax-mail header as the AAA account identifier.
gateway	Indicates that AAA uses the router-specific name derived from the host name and domain name as the AAA account identifier, displayed in the following format: *router-name.domain-name*.

Defaults

No account identification source is defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Normally, when AAA is being used for simple user authentication, AAA uses the username information defined in the user profile for authentication. With Store and Forward Fax, you can specify that the account ID, username, or recipient name from the e-mail header information be used to identify the user for authentication. This command defines what AAA uses for the secondary identifier for outbound or off-ramp user authentication with Store and Forward Fax.

Store and Forward Fax allows you to define either a primary or a secondary identifier. (You configure the secondary identifier using the **mmoip aaa send-id primary** command.) AAA extracts the authentication identifier information from the defined sources. If the field is blank (meaning undefined), AAA will use the secondary identifier source if configured. The secondary identifier is used only when the primary identifier is null. In this case, when AAA sees that the primary identifier is null, it will check to see if a secondary identifier has been defined and use that value for user authentication.

AAA does not use these methods sequentially—meaning that if the primary identifier is defined and AAA cannot match the primary identifier information, it will not use the secondary identifier for authentication. Authentication simply fails.

When you enable authentication, the on-ramp gateway inserts whatever value you configure for the **mmoip aaa receive-id secondary** command in the X-account-ID field of the e-mail header (if Store and Forward uses the defined secondary identifier). This X-account ID field contains the value that is used for authentication and accounting by the on-ramp gateway. For example, if the **mmoip aaa receive-id secondary** command is set to **gateway**, the on-ramp gateway name (for example, hostname.domain-name) is inserted in the X-account ID field of the e-mail header of the fax-mail message.

If you want to use this configured gateway value in the X-account ID field, you must configure the **mmoip aaa send-id secondary** command with the **account-id** keyword. This particular keyword enables Store and Forward Fax to generate end-to-end authentication and accounting tracking records. If you do not enable authentication on the on-ramp gateway, the X-account-ID field is left blank.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example defines the recipient name as defined in the envelope-to field of the e-mail header to be used as the AAA authentication identifier for Store and Forward Fax:

```
configure terminal
 aaa new-model
 mmoip aaa send-id secondary envelope-to
```

Related Commands

Command	Description
mmoip aaa send-id primary	Specifies the primary location where AAA retrieves its account identification information for off-ramp faxing.

mode (Voice over ATM)

To set the mode of the T1/E1 controller and enter specific configuration commands for each mode type, use the **mode** command in controller configuration mode. Use the **no** form of this command to restore the default mode of the controller.

mode {**atm** | **cas**}

no mode {**atm** | **cas**}

Syntax Description

atm	Sets the controller into ATM mode and creates an ATM interface (ATM 0) on the Cisco MC3810. When ATM mode is enabled, no channel groups, CAS groups, CCS groups, or clear channels are allowed because ATM occupies all the DS0s on the T1/E1 trunk. When you set the controller to ATM mode, the controller framing is automatically set to ESF for T1 or CRC4 for E1. The linecode is automatically set to B8ZS for T1 or HDBC for E1. When you remove ATM mode by entering the **no mode atm** command, ATM interface 0 is deleted. ATM mode is supported only on controller 0 (T1 or E1 0).
cas	Sets the controller into channel-associated signalling (CAS) mode, which allows you to create channel groups, CAS groups, and clear channels (both data and CAS modes). CAS mode is supported on both controllers 0 and 1.

Defaults

No mode is configured.

Command Modes

Controller configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to the Cisco MC3810 with the digital voice module (DVM) installed.

When no mode is selected, channel groups and clear channels (data mode) can be created using the **channel group** and **tdm-group** commands, respectively.

On the Cisco MC3810, some DS0s are used exclusively for different signalling modes. The DS0 channels have the following limitations when mixing different applications (such as voice and data) on the same network trunk:

- On E1 controllers, DS0 16 is used exclusively for either CAS or CCS, depending on which mode is configured.
- On T1 controllers, DS0 24 is used exclusively for CCS.

Examples

The following example configures ATM mode on controller T1 0. This step is required for Voice over ATM.

```
controller T1 0
 mode atm
```

The following example configures CAS mode on controller T1 1:

```
controller T1 1
 mode cas
```

Related Commands

Command	Description
channel-group	Defines the time slots that belong to each T1 or E1 circuit.
tdm-group	Configures a list of time slots for creating clear channel groups (pass-through) for TDM cross-connect.
voice-group	Configures a list of time slots for voice CAS on the T1/E1 controller on the Cisco MC3810 multiservice concentrator.

mode ccs

To configure the T1/E1 controller to support CCS cross-connect or CCS frame-forwarding, use the **mode ccs** command in controller configuration mode. To disable support for CCS cross-connect or CCS frame-forwarding on the controller, use the **no** form of this command.

mode ccs {**cross-connect** | **frame-forwarding**}

no mode ccs {**cross-connect** | **frame-forwarding**}

Syntax Description

cross-connect	Enables CCS cross-connect on the controller.
frame-forwarding	Enables CCS frame forwarding on the controller.

Defaults

No CCS mode is configured.

Command Modes

Controller configuration mode

Command History

Release	Modification
12.0(2)T	This command was introduced.

Examples

To enable CCS cross-connect on controller T1 1, enter the following commands:

```
controller T1 1
 mode ccs cross-connect
```

To enable CCS frame-forwarding on controller T1 1, enter the following commands:

```
controller T1 1
 mode ccs frame-forwarding
```

Related Commands

Command	Description
ccs connect	Configures a CCS connection on an interface configured to support CCS frame forwarding.

mta receive aliases

To specify a host name accepted as an SMTP alias for off-ramp faxing, use the **mta receive aliases** command in global configuration mode. Use the **no** form of this command to disable this alias.

mta receive aliases *string*

no mta receive aliases *string*

Syntax Description

string	Specifies the host name or IP address to be used as an alias for the SMTP server. If you specify an IP address to be used as an alias, you must enclose the IP address in brackets as follows: {xxx.xxx.xxx.xxx].

Defaults

Enabled with an empty string

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command creates an accept or reject alias list. The first alias is used by the mailer to identify itself in SMTP banners and when generating its own RFC 822 Received: header.

Note This command does not automatically include reception for a domain IP address—it must be explicitly added. To explicitly add a domain IP address, use the following format: **mta receive alias** [*ip-address*]. Use the IP address of the Ethernet and/or the FastEthernet interface of the off-ramp gateway.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example specifies the host name seattle-fax-offramp.example.com as the alias for the SMTP server:

```
configure terminal
 mta receive aliases seattle-fax-offramp.example.com
```

The following example specifies the host name 172.16.0.0 as the alias for the SMTP server:

```
configure terminal
 mta receive aliases [172.16.0.0]
```

Related Commands

Command	Description
mta receive generate-mdn	Specifies that the off-ramp gateway process a response MDN from an SMTP server.
mta receive maximum-recipients	Specifies the maximum recipients for all SMTP connections.

mta receive generate-mdn

To specify that the off-ramp gateway process a response message delivery notification (MDN) from an SMTP server, use the **mta receive generate-mdn** command in global configuration mode. Use the **no** form of this command to disable message delivery notice generation.

mta receive generate-mdn

no mta receive generate-mdn

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

When message delivery notification is enabled on a sending Cisco AS5300, the device inserts a flag in the off-ramp message e-mail header, requesting that the receiving Cisco AS5300 generate the message delivery notification and return that message to the sender when the e-mail message containing the fax image is opened. Use the **mta receive generate-mdn** command to enable the receiving device—the off-ramp gateway—to process the response message delivery notification.

Depending on the configuration, usage, and features of the mailers used at a site, it might be desirable to enable or disable MDN generation. (DSN generation cannot be disabled.)

Specifications for MDN are described in RFC 2298.

This command applies to off-ramp Store and Forward Fax functions.

Examples

The following example enables the receiving device to generate message delivery notices:

```
configure terminal
 mta receive generate-mdn
```

Related Commands

Command	Description
mdn	Requests that a message disposition notice be generated when the fax-mail message is processed (opened).
mta receive aliases	Specifies a host name accepted as an SMTP alias for off-ramp faxing.
mta receive maximum-recipients	Specifies the maximum recipients for all SMTP connections.

mta receive maximum-recipients

To specify the maximum recipients for all SMTP connections, use the **mta receive maximum-recipients** command in global configuration mode. Use the **no** form of this command to restore the default value.

mta receive maximum-recipients *number*

no mta receive maximum-recipients

Syntax Description

number	Specifies the maximum number of recipients for all SMTP connections. Valid entries are from 0 to 1024.

Defaults

The default is 0 recipients, meaning that incoming mail messages will not be accepted, thus no faxes will be sent by the off-ramp gateway.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command applies to off-ramp Store and Forward Fax functions.

Note Unless the sending mailer supports the X-SESSION SMTP service extension, each incoming SMTP connection will only be allowed to send to one recipient and thus only consume one outgoing modem.

Use the **mta receive maximum-recipients** command to configure the maximum number of modems you want to allocate for fax usage at any one time. You can use this command to limit the resource usage on the gateway. When the value for the *number* argument is set to 0, no new connections can be established. This is particularly useful when preparing to shut down the system.

Examples

The following example defines 10 as the maximum number of recipients for all SMTP connections:

```
configure terminal
 mta receive maximum-recipients 10
```

Related Commands

Command	Description
mta receive aliases	Specifies a host name accepted as an SMTP alias for off-ramp faxing.
mta receive generate-mdn	Specifies that the off-ramp gateway process a response MDN from an SMTP server.

mta send mail-from

To specify the mail-from address (also called the RFC 821 envelope-from or the Return-Path address), use the **mta send mail-from** command in global configuration mode. Use the **no** form of this command to disable this return path information.

mta send mail-from {**hostname** *string* | {**username** *string* | **username** s}}

no mta send mail-from {**hostname** *string* | {**username** *string* | **username** s}}

Syntax Description

hostname *string*	Text string that specifies the SMTP host name or IP address. If you specify an IP address, you must enclose the IP address in brackets as follows: {xxx.xxx.xxx.xxx].
username *string*	Text string that specifies the sender username.
username s	Wildcard that specifies that the username will be derived from the calling number.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Use the **mta send mail-from** command to designate the sender of the fax TIFF attachment. This value is equivalent to the return path information in an e-mail message.

The postmaster address, configured with the **mta send postmaster** command, is used if the mail-from address is blank.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example specifies that the mail-from username information will be derived from the sender's calling number:

```
configure terminal
 mta send mail-from username $s$
```

Related Commands

Command	Description
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.

Command	Description
mta send return-receipt-to	Specifies the address where MDNs will be sent.
mta send server	Specifies a destination mail server or servers.
mta send subject	Specifies the subject header of the e-mail message.

mta send origin-prefix

To add information to the e-mail prefix header, use the **mta send origin-prefix** command in global configuration mode. Use the **no** form of this command to disable the defined string.

mta send origin-prefix *string*

no mta send origin-prefix *string*

Syntax Description

string	Text string that adds comments to the e-mail prefix header. If this string contains more than one word, the string value should be contained within quotation marks ("x").

Defaults

Null string

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Normally, the Store and Forward Fax feature provides the slot and port number from which this e-mail came in the e-mail prefix header information. Use this command to append the defined text string to the front of the e-mail prefix header information. This test string is a prefix string that is appended with the modem port and slot number and passed in the originator_comment field of the esmtp_client_engine_open() call. Eventually, this ends up in the Received header field of the fax-mail message, for example:

```
Received (test onramp Santa Cruz slot1 port15) by router-5300.cisco.com for
<test-test@cisco.com> (with Cisco NetWorks); Fri, 25 Dec 1998 001500 -0800
```

In other words, using the command **mta send origin-prefix dog** will cause the Received header to contain the following information:

```
Received (dog, slot 3 modem 8) by as5300-sj.example.com.....
```

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example provides the user with additional information:

```
configure terminal
 mta send origin-prefix Cisco-Powered Fax System
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from or the Return-Path address).
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.
mta send return-receipt-to	Specifies the address where MDNs will be sent.
mta send server	Specifies a destination mail server or servers.
mta send subject	Specifies the subject header of the e-mail message.

mta send postmaster

To define where an e-mail should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination, use the **mta send postmaster** command in global configuration mode. Use the **no** form of this command to disable this defined postmaster.

mta send postmaster *e-mail-address*

no mta send postmaster *e-mail-address*

Syntax Description

e-mail-address	Defines where this e-mail should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

If you have configured the Cisco AS5300 to generate DSNs and MDNs but you have not configured the sender information (using the **mta send mail-from** command) or the SMTP server, DSNs and MDNs will be delivered to this e-mail address.

The address defined by this command is used as the mta send mail-from address if the evaluated string is blank. An address, such as fax-administrator@example.com, is recommended (where example.com is replaced with your domain name, and fax-administrator is aliased to the person responsible for the operation of the AS5300's fax functions). At some sites, this may be the same person as the e-mail postmaster, but at most sites this is likely to be a different person and thus should be a different e-mail address.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example configures the e-mail address fax-admin@example.com as the sender for all incoming faxes. Thus, any returned DSNs will be delivered to fax-admin@example.com if the Mail From filed is otherwise blank.

```
configure terminal
 mta send postmaster fax-admin@example.com
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from or the Return-Path address).
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send return-receipt-to	Specifies the address where MDNs will be sent.
mta send server	Specifies a destination mail server or servers.
mta send subject	Specifies the subject header of the e-mail message.

mta send return-receipt-to

To specify the address where MDNs will be sent, use the **mta send return-receipt-to** command in global configuration mode. Use the **no** form of this command to restore the default value.

mta send return-receipt-to {**hostname** *string* | {**username** *string* | **s**}}

no mta send return-receipt-to {**hostname** *string* | {**username** *string* | **s**}

Syntax Description

hostname *string*	Text string that specifies the SMTP host name where MDNs will be sent.
username *string*	Text string that specifies the sender's username where MDNs will be sent.
s	Wild card that specifies that the calling number (ANI) is used to generate the disposition-notification-to e-mail address.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Use the **mta send return-receipt-to** command to define where you want MDNs to be sent after the fax-mail is opened.

Note Store and Forward Fax supports Eudora's proprietary format, meaning that the header Store and Forward Fax generates is in compliance with RFC 2298 (MDN).

Note MMoIP dial peers must have MDN enabled to generate return receipts in off-ramp fax-mail messages.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example configures scoobee as the SMTP mail server to which DSNs will be sent:

```
configure terminal
 mta send return-receipt-to hostname server.com
 mta send return-receipt-to username scoobee
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from or the Return-Path address).
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.
mta send server	Specifies a destination mail server or servers.
mta send subject	Specifies the subject header of the e-mail message.

mta send server

To specify a destination mail server or servers, use the **mta send server** command in global configuration mode. Use the **no** form of this command to disable the specified destination mail server.

mta send server {*host-name* | *IP-address*}

no mta send server {*host-name* | *IP-address*}

Syntax Description

host-name	Defines the host name of the destination mail server.
IP-address	Defines the IP address of the destination mail server.

Defaults

IP address defined as 0.0.0.0

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

Use the **mta send server** command to provide a backup destination server in case the first configured mail server is unavailable. (This command is not intended to be used for load distribution.)

You can configure up to ten different destination mail servers using the **mta send server** command. If you configure more than one destination mail server, the Cisco AS5300 attempts to contact the first mail server configured. If that mail server is unavailable, it will contact the next configured destination mail server.

DNS MX records are not used to look up host names provided to this command.

Note When you use the **mta send server** command, you should configure the Cisco AS5300 to perform name lookups using the **ip name-server** command.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example defines the mail servers scoobee.example.com and doogie.example.com as the destination mail servers:

```
configure terminal
 mta send server scoobee.example.com
 mta send server doogie.example.com
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from or the Return-Path address).
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.
mta send return-receipt-to	Specifies the address where MDNs will be sent.
mta send subject	Specifies the subject header of the e-mail message.

mta send subject

To specify the subject header of the e-mail, use the **mta send subject** command in global configuration mode. Use the **no** form of this command to disable this string.

mta send subject *string*

no mta send subject *string*

Syntax Description

string	Text string that specifies the subject header of an e-mail message.

Defaults

Null string

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XJ	This command was introduced.

Usage Guidelines

This command applies to on-ramp Store and Forward Fax functions.

Note The string does not need to be enclosed in quotation marks.

Examples

The following example defines the subject header of an e-mail message as fax attachment:

```
configure terminal
 mta send subject fax attachment
```

Related Commands

Command	Description
mta send mail-from	Specifies the mail-from address (also called the RFC 821 envelope-from or the Return-Path address).
mta send origin-prefix	Adds information to the e-mail prefix header.
mta send postmaster	Defines where an e-mail message should be delivered (the mail server postmaster account) if it cannot be delivered to the defined destination.
mta send return-receipt-to	Specifies the address where MDNs will be sent.
mta send server	Specifies a destination mail server or servers.

music-threshold

To specify the threshold for on-hold music for a specified voice port, use the **music-threshold** command in voice-port configuration mode. Use the **no** form of this command to disable this feature.

music-threshold *number*

no music-threshold *number*

Syntax Description

number	The on-hold music threshold in decibels (dB). Valid entries are any integer from –70 to –30.

Defaults

–38 dB

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced for the Cisco 3600 series.
12.0(4)T	Support was added for the Cisco MC3810.

Usage Guidelines

Use this command to specify the decibel level of music played when calls are put on hold. This command tells the firmware to pass steady data above the specified level. It only affects the operation of VAD when receiving voice.

If the value for this command is set too high, VAD interprets music-on-hold as silence, and the remote end does not hear the music. If the value for this command is set too low, VAD compresses and passes silence when the background is noisy, creating unnecessary voice traffic.

Examples

The following example sets the decibel threshold for the music played when calls are put on hold to –35:

```
voice port 0:D
 music-threshold -35
```

The following example sets the decibel threshold to –35 for the music played when calls are put on hold on the Cisco 3600 series or Cisco MC3810:

```
voice-port 1/0/0
 music-threshold -35
```

Related Commands

Command	Description
voice-port	Opens voice-port configuration mode.

network-clock base-rate

To configure the network clock base rate for universal I/O serial ports 0 and 1 on the Cisco MC3810, use the **network-clock base-rate** command in global configuration mode. Use the **no** form of this command to disable the current network clock base rate.

network-clock base-rate {**56k** | **64k**}

no network-clock base-rate {**56k** | **64k**}

Syntax Description

56k	Sets the network clock base rate to 56 kilobits per second (kbps).
64k	Sets the network clock base rate to 64 kbps.

Defaults

56 kbps

Command Modes

Global configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Examples

The following example sets the network clock base-rate to 64 kbps:

```
network-clock base-rate 64k
```

Related Commands

Command	Description
network-clock-select (MC3810)	Uses the network clock source to provide timing to the system backplane PCM bus.
network-clock-switch	Configures the switch delay time to the next priority network clock source when the current network clock source fails.

network-clock-select (MC3810)

To use the network clock source to provide timing to the system backplane pulse code modulation (PCM) bus, use the **network-clock-select** command in global configuration mode. Use the **no** form of this command to cancel the network clock selection.

network-clock-select *priority* [**serial 0** | **system** | *controller*]

no network-clock-select *priority* [**serial 0** | **system** | *controller*]

Syntax Description

priority	Specifies the priority of the clock source.Valid entries are from 1 to 4. You can configure up to four clock sources. The higher the number of the clock source, the higher the priority. For example, clock source 1 has higher priority than clock source 2. When the higher priority clock source fails, after the delay specified using the **network-clock-switch** command, the next higher priority clock source is selected.
serial 0	(Optional) Specifies serial interface 0 as the clock source.
system	(Optional) Specifies the system clock as the clock source.
controller	(Optional) Specifies which controllers is the clock source. You can specify either the trunk controller (T1/E1 0) or the digital voice module (T1/E1/ 1).

Defaults

No network clock source is specified.

Command Modes

Global configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Examples

The following example sets the priority of four network clock sources. When the clock source with the highest priority (controller T1 0) fails, the Cisco MC3810 switches the clock source to the second highest priority (controller T1 1).

```
network-clock-select 1 T1 0
network-clock-select 2 T1 1
network-clock-select 3 serial 0
network-clock-select 4 System
```

Related Commands

Command	Description
network-clock-switch	Configures the switch delay time to the next priority network clock source when the current network clock source fails.

network-clock-switch

To configure the switch delay time to the next priority network clock source when the current network clock source fails, use the **network-clock-switch** command in global configuration mode. Use the **no** form of this command to cancel the network clock delay time selection.

network-clock-switch [*switch-delay* | **never**] [*restore-delay* | **never**]

no network-clock-switch *delay*

Syntax Description

switch-delay	(Optional) The delay time before the next priority network clock source is used when the current network clock source fails. The range is from 0 to 99 seconds. The default is 10 seconds.
never	(Optional) Indicates no delay time before the current network clock source recovers.
restore-delay	(Optional) The delay time before the current network clock source recovers. The range is from 0 to 99 seconds.
never	(Optional) Indicates no delay time before the next priority network clock source is used when the current network clock source fails.

Defaults

10 seconds

Command Modes

Global configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Examples

The following command switches the network clock source after 20 seconds and sets the delay time before the current network clock source recovers to 20 seconds:

```
network-clock-switch 20 20
```

Related Commands

Command	Description
network-clock-select (MC3810)	Uses the network clock source to provide timing to the system backplane PCM bus.

non-linear

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

non-linear

no non-linear

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

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

Enabling the **non-linear** command normally improves performance, although some users might perceive truncation of consonants at the end of sentences when this command is enabled.

Examples

The following example enables non-linear call processing on the Cisco 3600 series:

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

The following example enables non-linear call processing on the Cisco MC3810:

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

Related Commands

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

nsap

To specify the network service access point (NSAP) address for a local video dial peer, enter the **nsap** command in dial-peer configuration mode. Use the **no** form of the command to remove any configured NSAP address from the dial peer.

nsap *nsap-address*

no nsap

Syntax Description

nsap-address	Enter a 40-digit hexadecimal number; the number must be unique on the device.

Defaults

No video dial peer NSAP address is configured.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for ATM video dial-peer configuration on the Cisco MC3810.

Usage Guidelines

The address must be unique on the router.

Examples

On a Cisco MC3810, the following example sets up an NSAP address for the local video dial peer designated as 10:

```
dial-peer video 10 videocodec
 nsap 47.0091810000000002F26D4901.333333333332.02
```

Related Commands

Command	Description
dial-peer video	Defines a video ATM dial peer for a local or remote video codec, specifies video-related encapsulation, and enters dial-peer configuration mode.
show dial-peer video	Displays dial-peer configuration.

num-exp

To define how to expand an extension number into a particular destination pattern, use the **num-exp** command in global configuration mode. Use the **no** form of this command to cancel the configured number expansion.

num-exp *extension-number expanded-number*

no num-exp *extension-number expanded-number*

Syntax Description

extension-number	Digit(s) defining an extension number for a particular dial peer.
expanded-number	Digit(s) defining the expanded telephone number or destination pattern for the extension number listed.

Defaults

No number expansion is defined.

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **num-exp** global configuration command to define how to expand a particular set of numbers (for example, an extension number) into a particular destination pattern. With this command, you can map specific extensions and expanded numbers together by explicitly defining each number, or you can define extensions and expanded numbers using variables. You can also use this command to convert seven-digit numbers to numbers containing less than seven digits.

Use a period (.) as a variable or wild card, representing a single number. Use a separate period for each number you want to represent with a wildcard—meaning that if you want to replace four numbers in an extension with wildcards, type in four periods.

Examples

The following example expands the extension number 55541 to be expanded to 14085555541:

```
num-exp 65541 14085555541
```

The following example expands all five-digit extensions beginning with 5 to append the following numbers at the beginning of the extension number 1408555:

```
num-exp 5.... 1408555....
```

operation

To select a specific cabling scheme for E&M ports, use the **operation** command in voice-port configuration mode. Use the **no** form of this command to restore the default.

operation {**2-wire** | **4-wire**}

no operation {**2-wire** | **4-wire**}

Syntax Description

2-wire	Specifies a 2-wire E&M cabling scheme.
4-wire	Specifies a 4-wire E&M cabling scheme.

Defaults

2-wire operation

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to both the Cisco 3600 series and the Cisco MC3810.

The **operation** command only affects voice traffic. Signalling is independent of 2-wire versus 4-wire settings. If the wrong cable scheme is specified, the user might get voice traffic in only one direction.

Configuring the **operation** command on a voice port changes the operation of both voice ports on a VPM card. The voice port must be shut down and then opened again for the new value to take effect.

This command is not applicable to FXS or FXO interfaces because they are, by definition, 2-wire interfaces.

On the Cisco MC3810, this command only applies to the analog voice module (AVM).

Examples

The following example specifies that an E&M port on the Cisco 3600 series uses a 4-wire cabling scheme:

```
voice-port 1/0/0
 operation 4-wire
```

The following example specifies that an E&M port on the Cisco MC3810 uses a 2-wire cabling scheme:

```
voice-port 1/1
 operation 2-wire
```

output attenuation

To configure a specific output attenuation value, use the **output attenuation** command in voice-port configuration mode. Use the **no** form of this command to disable the selected output attenuation value.

output attenuation *value*

no output attenuation

Syntax Description

value	The amount of attenuation in decibels at the transmit side of the interface. Acceptable value is any integer from 0 to 14.

Defaults

The default value for FXO, FXS, and E&M ports is 0.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

A system-wide loss plan must be implemented using both **input gain** and **output attenuation** commands. Other equipment (including PBXs) in the system must be taken into account when creating a loss plan. This default value for this command assumes that a standard transmission loss plan is in effect, meaning that normally, there must be –6 dB attenuation between phones. Connections are implemented to provide –6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0.

Please note that you cannot increase the gain of a signal going out into the PSTN, but you can decrease it. Therefore, if the voice level is too high, you can decrease the volume by either decreasing the input gain value or by increasing the output attenuation.

You can increase the gain of a signal coming in to the router. If the voice level is too low, you can increase the input gain by using the **input gain** command.

Examples

The following example on the Cisco 3600 series configures a 3-decibel gain to be inserted at the transmit side of the interface:

```
voice-port 1/0/0
 output attenuation 3
```

The following example configures a 3-decibel gain on the Cisco AS5300 to be inserted at the transmit side of the interface:

```
voice-port 0:D
 output attenuation 3
```

The following example on the Cisco MC3810 configures a 6-decibel gain to be inserted at the transmit side of the interface:

```
voice-port 1/1
 output attenuation 6
```

Related Commands

Command	Description
input gain	Configures a specific input gain value for a voice port.

ping docsis

To determine whether a specific cable modem is online, use the **ping docsis** command in privileged EXEC mode.

ping docsis{*mac-addr* | *ip-addr*}

Syntax Description

mac-addr	MAC address. Specify the 48-bit hardware address of the cable modem.
ip-addr	IP address. Specify the IP address of the cable modem.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following example confirms that the cable modem at 172.00.00.00 is connected to the network and is operational:

```
ping docsis 172.00.00.00

Queueing 5 MAC-layer station maintenance intervals, timeout is 25 msec:
!!!!!
Success rate is 100 percent (5/5)
```

playout-delay

To tune the playout buffer on the Cisco MC3810 to accommodate packet jitter caused by switches in the WAN, use the **playout-delay** command in voice-port configuration mode. Use the **no** form of this command to restore the default value.

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

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

Syntax Description

maximum	Specifies the maximum playout delay. The maximum delay is the time the Cisco MC3810 digital signal processor (DSP) starts to discard voice packets.
nominal	Specifies the nominal playout delay. The nominal delay is the wait time that the Cisco MC3810 DSP starts to play out the voice packets.
value	The playout-delay value in milliseconds. The range for maximum playout delay is from 40 to 320, and the range for nominal playout delay is from 40 to 240.

Defaults

160 (maximum playout delay)
80 (nominal playout delay)

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies only to the Cisco MC3810.

Examples

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

```
voice-port 1/1
 playout-delay nominal 80
 playout-delay maximum 160
```

port (voice)

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 1750 router

port *slot-number/port*

no port *slot-number/port*

Cisco 2600/3600 series router

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

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

Cisco MC3810

port *slot/port*

no port *slot/port*

Cisco AS5300 access server

port *controller number*:**D**

no port *controller number*:**D**

Cisco AS5800 universal access server

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

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

Cisco 7200 series router

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

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

Cisco uBR924 cable access router

port *number*

no port *number*

Syntax Description

For the Cisco 1750 Router

slot-number	Slot number in the router where the VIC is installed. Valid entries are from 0 to 2, depending on the slot where it has been installed.
port	Indicates the voice port. Valid entries are 0 or 1.

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.
slot	Router location where the voice port adapter is installed. Valid entries are from 0 to 3.
port	Voice interface card location. Valid entries are 0 or 3.
dso-group-no	Indicates the defines DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

For the Cisco MC3810:

slot/port	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 T1: from 1 to 24. Digital E1: from 1 to 15, and from 17 to 31.

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.

For the Cisco 7200 Series Router

slot	Router location where the voice port adapter is installed. Valid entries are from 0 to 3.
port	Voice interface card location. Valid entries are 0 or 1.

dso-group-no	Indicates the defines DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.
slot-number	Indicates the 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	Indicates the subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1.
port	Indicates the voice port number. Valid entries are 0 or 1.

For the Cisco uBR924 cable access router

number	Indicates the RJ-11 connectors installed in the Cisco uBR924. Valid entries are 0 (which corresponds to the RJ-11 connector labeled V1) and 1 (which corresponds to the RJ-11 connector labeled V2.

Defaults

No port is configured.

Command Modes

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.
11.3MA	Port-specific values for the Cisco MC3810 were added.
12.0(3)T	Port-specific values for the Cisco AS5300 were added.
12.0(4)T	Support was added for the Cisco uBR924 platform.
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.

Examples

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
```

port media

To specify the serial interface where the local video codec is connected for a local video dial peer, use the **port media** command in video dial-peer configuration configuration mode. Use the **no** form of the command to remove any configured locations from the dial peer.

port media *interface*

no port media

Syntax Description

interface	Indicates the serial interface where the local codec is connected. Valid entries are the numbers 1 or 0.

Defaults

No interface is specified.

Command Modes

Video dial-peer configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for ATM video dial-peer configuration on the Cisco MC3810.

Examples

On a Cisco MC3810 local video dial peer designated as 10, the following example shows serial interface 0 as the specified interface for the codec:

```
dial-peer video 10 videocodec
 port media Serial0
```

Related Commands

Command	Description
port signal	Specifies the slot location of the VDM and the port location of the EIA/TIA-366 interface for signalling.
show dial-peer video	Displays dial-peer configuration.

port signal

To specify the slot location of the video dialing module (VDM) and the port location of the RS-366 interface for signalling for a local video dial peer, use the **port signal** command in video dial-peer configuration mode. Use the **no** form of the command to remove any configured locations from the dial peer.

port signal *slot/port*

no port signal

Syntax Description

slot	Enter either 1 or 2 as the slot location of the VDM.
port	Enter the port location of the RS-366 interface. The Cisco MC3810 VDM has only one port, so the *port* value is always 0.

Defaults

No locations are specified.

Command Modes

Video dial-peer configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for ATM video dial-peer configuration on the Cisco MC3810.

Examples

On a Cisco MC3810, the following example shows how to set up the VDM and RS-366 interface locations for the local video dial peer designated as 10:

```
dial-peer video 10 videocodec
 port signal 1/0
```

Related Commands

Command	Description
port media	Specifies the serial interface where the local video codec is connected.
show dial-peer video	Displays dial-peer configuration.

pots country

To configure your connected telephones, fax machines, or modems to use country-specific default settings for each physical characteristic, use the **pots country** command in global configuration mode. Use the **no** form of this command to disable the use of country-specific default settings for each physical characteristic.

pots country *country*

no pots country *country*

Syntax Description

country	Country that your router is in. Enter the **pots country ?** command to get a list of supported countries and the code you must enter to indicate a particular country.

Defaults

A default country is not defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to the Cisco 800 series routers.

If you need to change a country-specific default setting of a physical characteristic, you can use the associated command listed in the "Related Commands" section.

Examples

The following example specifies that the devices connected to the telephone ports use default settings specific to Germany for the physical characteristics:

```
pots country de
```

Related Commands

Command	Description
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).

Command	Description
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots dialing-method

To specify how the router collects and sends digits dialed on your connected telephones, fax machines, or modems, use the **pots dialing-method** command in global configuration mode. Use the **no** form of this command to disable the specified dialing method.

pots dialing-method {**overlap** | **enblock**}

no pots dialing-method {**overlap** | **enblock**}

Syntax Description

overlap	The router sends each digit dialed in a separate message.
enblock	The router collects all digits dialed and sends the digits in one message.

Defaults

Depends on the setting of the **pots country** command. For more information, refer to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 800 series routers.

To interrupt the collection and transmission of dialed digits, enter a pound sign (#) or stop dialing digits until the interdigit timer runs out (10 seconds).

Examples

The following example specifies that the router uses the enblock dialing method:

```
pots dialing-method enblock
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).

Command	Description
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots disconnect-supervision

To specify how a router notifies the connected telephones, fax machines, or modems when the calling party has disconnected, use the **pots disconnect-supervision** command in global configuration mode. Use the **no** form of this command to disable the specified disconnect method.

pots disconnect-supervision {**osi** | **reversal**}

no pots disconnect-supervision {**osi** | **reversal**}

Syntax Description

osi	Open switching interval (OSI) is the duration for which DC voltage applied between tip and ring conductors of a telephone port is removed.
reversal	Polarity reversal of tip and ring conductors of a telephone port.

Defaults

Depends on the setting of the **pots country** command. For more information, refer to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 800 series routers.

Most countries except Japan typically use the **osi** option. Japan typically uses the **reversal** option.

Examples

The following example specifies that the router uses the osi disconnect method:

```
pots disconnect-supervision osi
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).

Command	Description
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots disconnect-time

To specify the uuuuuuuuuuiiiinterval in which the disconnect method is applied if your connected telephones, fax machines, or modems fail to detect that a calling party has disconnected, use the **pots disconnect-time** command in global configuration mode. Use the **no** form of this command to disable the specified disconnect interval.

pots disconnect-time *interval*

no pots disconnect-time *interval*

Syntax Description

interval	Number from 50 to 2000 (milliseconds).

Defaults

Depends on the setting of the **pots country** command. For more information, refer to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

The **pots disconnect-supervision** command configures the disconnect method.

Examples

This command applies to Cisco 800 series routers.

The following example specifies that the connected devices apply the configured disconnect method for 100 milliseconds after a calling party disconnects:

```
pots disconnect-time 100
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).

Command	Description
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots distinctive-ring-guard-time

To specify a idelay in which a telephone port can be rung after a previous call is disconnected, use the **pots distinctive-ring-guard-time** command in global configuration mode. Use the **no** form of this command to disable the specified delay.

pots distinctive-ring-guard-time *milliseconds*

no pots distinctive-ring-guard-time *milliseconds*

Syntax Description

milliseconds	Number from 0 to 1000 (milliseconds).

Defaults

Depends on the setting of the **pots country** command. For more information, refer to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example specifies that a telephone port can be rung 100 milliseconds after a previous call is disconnected:

```
pots distinctive-ring-guard-time 100
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.

Command	Description
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
ring	Sets up a distinctive ring for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots encoding

To specify the pulse code modulation (PCM) encoding scheme for your connected telephones, fax machines, or modems, use the **pots encoding** command in global configuration mode. Use the **no** form of this command to disable the specified PCM encoding scheme.

pots encoding {**alaw** | **ulaw**}

no pots encoding {**alaw** | **ulaw**}

Syntax Description

alaw	International Telecommunication Union Telecommunication Standardization Section (ITU-T) PCM encoding scheme used to represent analog voice samples as digital values.
ulaw	North American PCM encoding scheme used to represent analog voice samples as digital values.

Defaults

Depends on the setting of the **pots country** command. For more information, refer to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 800 series routers.

Europe typically uses the **alaw** option. North America typically uses the **ulaw** option.

Examples

The following example specifies **alaw** as the PCM encoding scheme:

```
pots encoding alaw
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.

Command	Description
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots line-type

To specify the impedance of your connected telephones, fax machines, or modems, use the **pots line-type** command in global configuration mode. Use the **no** form of this command to disable the specified line type.

pots line-type {**type1** | **type2** | **type3**}

no pots line-type {**type1** | **type2** | **type3**}

Syntax Description

type1	Runs at 600 ohms.
type2	Runs at 900 ohms.
type3	Runs at 300/400 ohms.

Defaults

Depends on the setting of the **pots country** command. For more information, refer to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example specifies type1 as the line type:

```
pots line-type type1
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.

Command	Description
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots ringing-freq

To specify the frequency at which your connected telephones, fax machines, or modems ring, use the **pots ringing-freq** command in global configuration mode. Use the **no** form of this command to disable the specified ringing frequency.

Cisco 800 series router

pots ringing-freq {**20Hz** | **25Hz** | **50Hz**}

no pots ringing-freq {**20Hz** | **25Hz** | **50Hz**}

Syntax Description

20Hz	Connected devices ring at 20 Hz.
25Hz	Connected devices ring at 25 Hz.
50Hz	Connected devices ring at 50 Hz.

Defaults

Depends on the setting of the **pots country** command. For more information, refer to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example specifies a ringing frequency of 50 Hz:

```
router (config)# pots ringing-freq 50Hz
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.

Command	Description
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots silence-time

To specify the interval of silence after a calling party disconnects, use the **pots silence-time** command in global configuration mode. Use the **no** form of this command to disable the specified silence time.

pots silence-time *interval*

no pots silence-time *interval*

Syntax Description

interval	Number from 0 to 10 (seconds).

Defaults

Depends on the setting of the **pots country** command. For more information, refer to the **pots country** command.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 800 series routers.

Examples

The following example specifies 10 seconds as the interval of silence:

```
pots silence-time 10
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.

Command	Description
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pots tone-source

To specify the source of dial, ringback, and busy tones for your connected telephones, fax machines, or modems, use the **pots tone-source** command in global configuration mode. Use the **no** form of this command to disable the specified tone source.

pots tone-source {**local** | **remote**}

no pots tone-source {**local** | **remote**}

Syntax Description

local	Router supplies the tones.
remote	Telephone switch supplies the tones.

Defaults

The default setting is **local**.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 800 series routers.

This command applies only to ISDN lines connected to a EURO-ISDN (NET3) switch.

Examples

The following example specifies **remote** as the tone source:

```
pots tone-source remote
```

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).

Command	Description
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
show pots status	Displays the settings of the telephone port physical characteristics and other information on the telephone interfaces on a Cisco 800 series router.

pre-dial delay

To configure a delay on an Foreign Exchange Office (FXO) interface between the beginning of the off-hook state and the initiation of dual-tone multifrequency (DTMF) signalling, use the **pre-dial delay** voice-port configuration command. The **no** form of the command restores the default value.

pre-dial delay *seconds*

no pre-dial delay

Syntax Description

seconds	The delay before signalling begins. Valid values are from 0 to 10.

Defaults

1 second.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.(7)T and 12.0(2)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 3600 series routers.

To disable the command, set the delay to 0.

When an FXO interface begins to draw loop current (off-hook state), a delay is required between the initial flow of loop current and the beginning of signalling. Some devices initiate signalling too quickly, resulting in redial attempts. The **pre-dial delay** command allows a signalling delay.

Examples

The following example sets a pre-dial delay value of 3 seconds on a Cisco 3600 series router FXO port:

```
voice-port 1/0/0
 pre-dial delay 3
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit time-out value for a specified voice port.
timing delay-duration	Configures delay dial signal duration for a specified voice port.

preference

To indicate the preferred order of a dial peer within a hunt group, use the **preference** command in dial-peer configuration mode. Use the **no** form of this command to remove the preference value on the voice port.

preference *value*

no preference *value*

Syntax Description

value	An integer from 0 to 10, where the lower the number, the higher the preference. The default value is 0 (highest preference).

Defaults

0 (highest preference)

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(3)T	This command was supported on the Cisco 2600 series and 3600 series routers.
12.0(4)T	Support was added for VoFR dial peers on the Cisco 2600 series and 3600 series routers.

Usage Guidelines

This command applies to POTS dial peers, Voice over IP (VoIP) dial peers, and Voice over Frame Relay (VoFR) dial peers. This command applies to POTS dial peers, VoFR dial peers, Voice over ATM dial peers, and Voice over HDLC dial peers on the Cisco MC3810.

Use the **preference** command to indicate the preference order for matching dial peers in a rotary group. Setting the preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

Note If POTS and voice-network peers are mixed in the same hunt group, the POTS dial peers must have priority over the voice-network dial peers.

Use this command with the Rotary Calling Pattern feature.

The hunting algorithm precedence is configurable. For example, if you wish a call processing sequence to go to destination #A first, then destination B second, and third to destination C; you would assign preference (0 being the highest priority) to the destinations in the following order:

- Preference 0 to A
- Preference 1 to B
- Preference 2 to C

Examples

The following example configures POTS dial peer 10 to a preference of 1, POTS dial peer 20 to a preference of 2, and VoFR dial peer 30 to a preference of 3:

```
dial-peer voice 10 pots
 destination pattern 5552150
 preference 1
 exit

dial-peer voice 20 pots
 destination pattern 5552150
 preference 2
 exit

dial-peer voice 30 vofr
 destination pattern 5552150
 preference 3
 exit
```

The following examples show different dial peer configurations using the **preference** command:

Example 1

```
Dialpeer        destpat         preference          session-target
1               4085551048      0 (highest)         jmmurphy-voip
2               408555          0                   sj-voip
3               408555          1 (lower)           backup-sj-voip
4               ..........      1                   0:D      (interface)
5               ..........      0                   anywhere-voip
```

If the destination number is 4085551048, the order of attempts will be 1, 2, 3, 5, 4:

Example 2

```
Dialpeer        destpat         preference
1               408555          0
2               4085551048      1
3               4085551         0
4 ..............4085551.........0
```

The number dialed is 4085551048, the order will be 2, 3, 4, 1.

Note The default behavior is that the longest matching dial peer supersedes the preference value.

Related Commands

Command	Description
called-number (dial-peer)	Enables an incoming VoFR call leg to get bridged to the correct POTS call leg when using a static FRF.11 trunk connection.
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
cptone	Specifies a regional analog voice interface-related tone, ring, and cadence setting.
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.

Command	Description
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
signal-type	Sets the signalling type to be used when connecting to a dial peer.

prefix

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

prefix *string*

no prefix

Syntax Description

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

Defaults

Null string

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.

Usage Guidelines

This command is applicable only to POTS dial peers. This command applies to off-ramp Store and Forward Fax functions.

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

If you want to configure different prefixes for dialed numbers on the same interface, you need to configure different dial peers.

Examples

The following example specifies a prefix of 9 and then a pause:

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

Related Commands

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

pri-group nec-fusion

To configure your NEC PBX to support Fusion Call Control Signalling (FCCS), use the **pri-group nec-fusion** command in controller configuration mode. To disable FCCS, use the **no** form of this command.

pri-group nec-fusion {*pbx-ip-address* | *pbx-ip-host-name*} **pbx-port** *number*

no pri-group nec-fusion {*pbx-ip-address* | *pbx-ip-host-name*} **pbx-port** *number*

Syntax Description

pbx-ip-address	The IP address of the NEC PBX.
pbx-ip-host-name	The host name of the NEC PBX.
number	Choose a port number for the PBX. The range for the PBX port is 49152 to 65535. If you don't specify a port number, the default value of 55000 will be used. If this value is already in use, the next greater value will be used.

Defaults

55000

Command Modes

Controller configuration

Command History

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

Usage Guidelines

This command is used only if the PBX in your configuration is an NEC PBX, and if you are configuring it to run FCCS and not Q.SIG signalling.

Examples

The following example shows how to configure this NEC PBX to use FCCS:

```
pri-group nec-fusion 172.31.255.255 pbx-port 60000
```

Related Commands

Command	Description
isdn protocol-emulate	Configures the Layer 2 and Layer 3 port protocol of a BRI voice port or a PRI interface to emulate NT (network) or TE (user) functionality.
isdn switch type	Configures the Cisco AS5300 PRI interface to support Q.SIG signalling.
show cdapi	Displays the CDAPI.
show rawmsg	Displays the raw messages owned by the required component.

proxy h323

To enable the proxy feature on your router, use the **proxy h323** command in global configuration mode. To disable the proxy feature, use the **no** form of this command.

proxy h323

no proxy h323

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced.

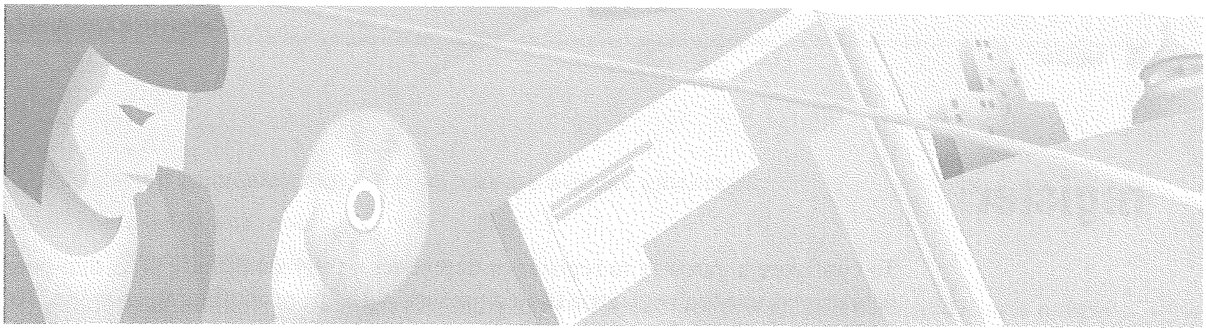
Usage Guidelines

Note If the multimedia interface is not enabled using the **proxy h323** command, or if no gatekeeper is available, starting the proxy allows it to attempt to locate these resources. No calls will be accepted until the multimedia interface and the gatekeeper are found.

Examples

The following example turns on the proxy feature:

```
proxy h323
```



Multiservice Applications Commands: R through Sh

This book documents commands used to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features. Commands in this book are listed alphabetically. For information on how to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features, refer to the *Cisco IOS Multiservice Applications Configuration Guide*.

register

To configure a gateway to register or deregister a fully-qualified POTS dial-peer E.164 address with a gatekeeper, use the **register e164** command in dial peer configuration mode. To deregister an E.164 address, use the **no** form of this command.

register e164

no register e164

Syntax Description

This command has no keywords or arguments.

Defaults

No E.164 addresses are registered until you enter this command.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(5)T	This command was introduced.

Usage Guidelines

Use this command to register the E.164 address of an analog telephone line attached to an FXS port on a router. The gateway automatically registers fully-qualified E164 addresses. Use the **no register e164** command to deregister an address. Use the **register e164** command to register a deregistered address.

Before you automatically or manually register an E.164 address with a gatekeeper, you must create a dial peer (**dial-peer** command), assign an FXS port to the peer (**port** command), and assign an E.164 address by using the **destination-pattern** command. The E.164 address must be a fully-qualified address. For example, +5551212, 5551212, and 4085551212 are fully-qualified addresses; 408555.... is not a fully-qualified address. E.164 addresses are only registered for active interfaces—those that are not shut down. If an FXS port or its interface is shut down, the corresponding E.164 address is deregistered.

Tips You can use the show gateway command to find out if the gateway is connected to a gatekeeper and if a fully-qualified E.164 address is assigned to the gateway. Use the **zone-prefix** command at the gatekeeper to define prefix patterns, such as 408555...., that apply to one or more gateways.

Examples

The following command sequence places the gateway in dial-peer configuration mode, assigns a E.164 address to the interface, and registers that address with the gatekeeper:

```
dial-peer voice 111 pots
port 1/0/0
destination-pattern 5551212
register e164
```

The following commands deregister an address with the gatekeeper:

```
dial-peer voice 111 pots
no register e164
```

The following example shows that you must have a connection to a gatekeeper and define a unique E.164 address before you can register an address:

```
dial-peer voice 222 pots
port 1/0/0
destination 919555....
register e164


ERROR-register-e164:Dial-peer destination-pattern is not a full E.164 number

no gateway
dial-peer voice 111 pots
register e164


ERROR-register-e164:No gatekeeper
```

Related Commands

Command	Description
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dial-peer	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
port	Enables an interface on a PA-4R-DTR port adapter to operate as a concentrator port.
show gateway	Displays the current gateway status.
zone prefix	Configures the gatekeeper with knowledge of its own prefix and the prefix of any remote zone.

resource threshold

To configure a gateway to report H.323 resource availability to the its gatekeeper, use the **resource threshold** command in the gateway configuration mode. To disable gateway resource-level reporting, use the **no** form of this command.

resource threshold [**all**] [**high** *percentage-value*] [**low** *percentage-value*]

no resource threshold

Syntax Description

all	(Optional) Applies the high- and low- parameter settings to all monitored H.323 resources. This is the default condition.
high *percentage-value*	(Optional) A resource utilization level that triggers a Resource Availability Indicator (RAI) message that indicates that H.323 resource use is high. Enter a number between 1 and 100 that represents the high-resource utilization percentage. A 100 value specifies high-resource usage when any H.323 resource is unavailable. The default is 90 percent.
low *percentage-value*	(Optional) Resource utilization level that triggers an RAI message that indicates that H.323 resource usage has dropped below the high usage level. Enter a number between 1 and 100 that represents the acceptable resource utilization percentage. After the gateway sends a high-utilization message, it waits to send the resource recovery message until the resource use drops below the value defined by the **low** parameter. The default is 90 percent.

Defaults

Reports low resources when 90 percent of resources are in use, and reports resource availability when resource use drops below 90 percent.

Command Modes

Gateway configuration

Command History

Release	Modification
12.0(5)T	This command was introduced.

Usage Guidelines

The **resource threshold** command defines the resource load levels that trigger Resource Availability Indicator (RAI) messages. To view the monitored resources, enter the **show gateway** command.

The monitored H.323 resources include digital signal processor (DSP) channels and DS0s. Use the **show call resource voice stats** command to see the total amount of resources available for H.323 calls.

Note The DS0 resources that are monitored for H.323 calls are limited to the ones that are associated with a voice POTS dial peer.

See the **dial-peer** configuration commands for details on how to associate a dial peer with a PRI or CAS group.

When any monitored H.323 resources exceed the threshold level defined by the **high** parameter, the gateway sends an RAI message to the gatekeeper with the AlmostOutOfResources field flagged. This message reports high-resource usage.

When all gateway H.323 resources drop below the level defined by the **low** parameter, the gateway sends the RAI message to the gatekeeper with the AlmostOutOfResources field cleared.

When a gatekeeper can choose between multiple gateways for call completion, the gatekeeper uses internal priority settings and gateway resource statistics to determine which gateway to use. When all other factors are equal, a gateway that has available resources will be chosen over a gateway that has reported limited resources.

Examples

The following command defines the H.323 resource limits for a gateway:

```
resource threshold high 70 low 60
```

Related Commands

Command	Description
show gateway	Displays the current gateway status.
show call resource voice stats	Displays resource statistics for an H.323 gateway.
show call resource voice threshold	Displays the threshold configuration settings and status for an H.323 gateway.

req-qos

To specify the desired quality of service to be used in reaching a specified dial peer, use the **req-qos** command in dial-peer configuration mode. To restore the default value for this command, use the **no** form of this command.

req-qos {**best-effort** | **controlled-load** | **guaranteed-delay**}

no req-qos

Syntax Description

best-effort	Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation.
controlled-load	Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to assure that preferential service is received even when the bandwidth is overloaded.
guaranteed-delay	Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queueing if the bandwidth reserved is not exceeded.

Defaults

best-effort

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command is applicable only to VoIP dial peers.

Use the **req-qos** command to request a specific quality of service to be used in reaching a dial peer. Like **acc-qos**, when you issue this command, the Cisco IOS software reserves a certain amount of bandwidth so that the selected quality of service can be provided. Cisco IOS software uses Resource Reservation Protocol (RSVP) to request quality of service guarantees from the network.

Examples

The following example configures guaranteed-delay as the desired (requested) quality of service to a dial peer:

```
dial-peer voice 10 voip
 req-qos guaranteed-delay
```

Related Commands

Command	Description
acc-qos	Generates an SNMP event if the quality of service for a dial peer drops below a specified level.

reset

To reset a set of digital signal processor (DSP)s, use the **reset** command in global configuration mode.

reset *number*

Syntax Description

number	Specifies the number of DSPs to be reset. The number of DSPs range from 0 to 30.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

12.0(5)XE and 12.0(7)T	This command was introduced.

Examples

The following example displays the **reset** command configuration for DSP 1:

```
reset 1
01:24:54:%DSPRM-5-UPDOWN: DSP 1 in slot 1, changed state to up
```

response-timeout

To configure the maximum time to wait for a response from a server, use the **response-timeout** command in settlement configuration mode. To reset to the default value of this command, use the **no** form of this command.

response-timeout *num*

no response-timeout *num*

Syntax Description

num	Indicates the response waiting time in seconds.

Defaults

The default response timeout is 1 second (one second).

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

If no response is received within this time limit, the current connection ends and the router attempts to contact the next service point.

Examples

The following example configures a 1-second time to wait for a response from a server:

```
settlement 0
 response-timeout 1
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of connection attempts to the provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.

Command	Description
settlement	Enters settlement mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

retry-delay

To set the time between attempts to connect with the settlement provider, use the **retry-delay** command in settlement configuration mode. To reset to the default value of this command, use the **no** form of this command.

retry-delay *num*

no retry-delay

Syntax Description

num	Length of time (in seconds) between attempts to connect with the settlement provider. The valid range for retry-delay is 1 to 600 seconds.

Defaults

The default retry delay is 2 seconds.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

To set the time between attempts to connect with the settlement provider, use the **retry-delay** command in the settlement configuration mode. After exhausting all service points for the provider, the router is delayed for this length of time before resuming connection attempts.

Examples

The following example sets a retry value of 15 seconds:

```
settlement 0
 relay-delay 15
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-limit	Sets the maximum number of connection attempts to the provider.

Command	Description
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

retry-limit

To set the maximum number of connection attempts to the provider, use the **retry-limit** command in settlement configuration mode. To reset to the default value of this command, use the **no** form of this command.

retry-limit *num*

no retry-limit *num*

Syntax Description

num	Maximum number of connection attempts in addition to the first attempt.

Defaults

The default retry limit is one (1) retry.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

If no connection is established after the configured retries, the router ceases connection attempts. The retry limit number does not count the initial connection attempt. A retry limit of one (default) results in a total of two connection attempts to every service point.

Examples

The following example sets the number of retries to 1:

```
settlement 0
 relay-limit 1
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.

Command	Description
settlement	Enters settlement mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

ring

To set up a distinctive ring for your connected telephones, fax machines, or modems, use the **ring** command in interface configuration mode. To disable the specified distinctive ring, use the **no** form of this command.

ring *cadence-number*

no ring *cadence-number*

Syntax Description

cadence-number	Number from 0 through 2: • Type 0 is a primary ringing cadence—default ringing cadence for country your router is in. • Type 1 is a distinctive ring—0.8 seconds on, 0.4 seconds off, 0.8 seconds on, 0.4 seconds off. • Type 2 is a distinctive ring—0.4 seconds on, 0.2 seconds off, 0.4 seconds on, 0.2 seconds off, 0.8 seconds on, 4 seconds off.

Defaults

The default is 0.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 800 series routers.

You can specify this command when creating a dial peer. This command will not work if it is not specified within the context of a dial peer. For information on creating a dial peer, refer to the *Cisco 800 Series Routers Software Configuration Guide*.

Examples

The following example specifies the type 1 distinctive ring:

```
ring 1
```

Related Commands

Command	Description
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
no call-waiting	Disables call waiting.
port (dial-peer)	Enables an interface on a PA-4R-DTR port adapter to operate as a concentrator port.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
ring	Sets up a distinctive ring for telephones, fax machines, or modems connected to a Cisco 800 series router.
show dial-peer voice	Displays configuration information and call statistics for dial peers.

ring cadence

To specify the ring cadence for an FXS voice port on the Cisco MC3810, use the **ring cadence** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

ring cadence [**on1** | **off1**] [**on2** | **off2**] [**on3** | **off3**] [**on4** | **off4**] [**on5** | **off5**] [**on6** | **off6**]

no ring cadence

Syntax Description

on1	(Optional) Pulses on for 100 milliseconds.
off1	(Optional) Pulses off for 100 milliseconds.
on2	(Optional) Pulses on for 200 milliseconds.
off2	(Optional) Pulses off for 200 milliseconds.
on3	(Optional) Pulses on for 300 milliseconds.
off3	(Optional) Pulses off for 300 milliseconds.
on4	(Optional) Pulses on for 400 milliseconds.
off4	(Optional) Pulses off for 400 milliseconds.
on5	(Optional) Pulses on for 500 milliseconds.
off5	(Optional) Pulses off for 500 milliseconds.
on6	(Optional) Pulses on for 600 milliseconds.
off6	(Optional) Pulses off for 600 milliseconds.

Defaults

on2 off4 (default North American ring pattern)

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies only to the Cisco MC3810.

Examples

The following example configures the ring cadence for 0.4 seconds on and 0.2 seconds off on voice port 1/1 on the Cisco MC3810:

```
voice-port 1/1
 ring cadence on4 off2
```

Related Commands

Command	Description
ring frequency	Specifies the ring frequency for a specified FXS voice port.
ring number	Specifies the number of rings for a specified FXO voice port.

ring frequency

To specify the ring frequency for a specified FXS voice port, use the **ring frequency** command in voice-port configuration mode. To restore the default value for this command, use the **no** form of this command.

ring frequency *number*

no ring frequency *number*

Syntax Description

number	Ring frequency (hertz) used in the FXS interface. Valid entries on the Cisco 3600 series are 25 and 50. Valid entries on the Cisco MC3810 are 20 and 30.

Defaults

25 Hz on the Cisco 3600 series and 20 Hz on the Cisco MC3810

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **ring frequency** command to select a specific ring frequency for an FXS voice port. Use the **no** form of this command to reset the default value for this command. The ring frequency you select must match the connected equipment. If set incorrectly, the attached phone might not ring or might buzz. In addition, the ring frequency is usually country-dependent and you should take into account the appropriate ring frequency for your area before configuring this command.

This command does not affect ringback, which is the ringing a user hears when placing a remote call.

Examples

The following example configures the ring frequency on the Cisco 3600 series for 25 Hz:

```
voice-port 1/0/0
 ring frequency 25
```

The following example configures the ring frequency on the Cisco MC3810 for 20 Hz:

```
voice-port 1/1
 ring frequency 20
```

Related Commands

Command	Description
ring cadence	Specifies the ring cadence for an FXS voice port on the Cisco MC3810 multiservice concentrator.
ring number	Specifies the number of rings for a specified FXO voice port.

ring number

To specify the number of rings for a specified FXO voice port, use the **ring number** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

ring number *number*

no ring number *number*

Syntax Description

number	Number of rings detected before answering the call. Valid entries are numbers from 1 to 10. The default is 1.

Defaults

One ring

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **ring number** command to set the maximum number of rings to be detected before answering a call over an FXO voice port. Use the **no** form of this command to reset the default value, which is one ring.

Normally, this command should be set to the default so that incoming calls are answered quickly. If you have other equipment available on the line to answer incoming calls, you might want to set the value higher to give the equipment sufficient time to respond. In that case, the FXO interface would answer if the equipment on line did not answer the incoming call in the configured number of rings.

This command is not applicable to FXS or E&M interfaces because they do not receive ringing to receive a call.

Examples

The following example on the Cisco 3600 series sets five rings as the maximum number of rings to be detected before closing a connection over this voice port:

```
voice-port 1/0/0
 ring number 5
```

The following example on the Cisco MC3810 sets five rings as the maximum number of rings to be detected before closing a connection over this voice port:

```
voice-port 1/1
 ring number 5
```

Related Commands

Command	Description
ring frequency	Specifies the ring frequency for a specified FXS voice port.

security

To enable authentication and authorization on a gatekeeper, use the **security** command in gatekeeper configuration mode. To disable security, use the **no** form of this command.

security {**any** | **h323-id** | **e164**} {**password default** *password* | **password separator** *character*}

no security {**any** | **h323-id** | **e164**} {**password default** *password* | **password separator** *character*}

Syntax Description

any	Uses the first alias of an incoming RAS registration, regardless of its type, as the means of identifying the user to RADIUS/TACACS+.
h323-id	Uses the first H.323 ID type alias as the means of identifying the user to RADIUS/TACACS+.
e164	Uses the first E.164 address type alias as the means of identifying the user to RADIUS/TACACS+.
password default *password*	Specifies the default password that the gatekeeper associates with endpoints when authenticating them with an authentication server. The *password* must be identical to the password on the authentication server.
password separator *character*	Specifies the character that endpoints use to separate the H.323-ID from the piggybacked password in the registration. This allows each endpoint to supply a user-specific password. The separator character and password will be stripped from the string before it is treated as an H.323-ID alias to be registered. Note that passwords may only be piggybacked in the H.323-ID, not the E.164 address. This is because the E.164 address allows a limited set of mostly numeric characters. If the endpoint does not wish to register an H.323-ID, it can still supply an H.323-ID consisting of just the separator character and password. This will be understood to be a password mechanism and no H.323-ID will be registered.

Defaults

Disabled

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced.

Usage Guidelines

Use the security command to enable identification of registered aliases by RADIUS/TACACS+. If the alias does not exist in RADIUS/TACACS+, the endpoint will not be allowed to register.

A RADIUS/TACACS+ server and encryption key must have been configured in Cisco IOS software for security to work.

Only the first alias of the proper type will be identified. If no alias of the proper type is found, the registration will be rejected.

This command does not allow you to define the password mechanism unless the security type (**h323-id** or **e164** or **any**) has been defined. While the **no security password** command undefines the password mechanism, it leaves the security type unchanged, so security is still enabled. However, the **no security** {**h323-id** | **e164** | **any**} command disables security entirely, including removing any existing password definitions.

Examples

The following example enables identification of registrations using the first H.323 ID found in any registration:

```
security h323id
```

The following example enables security, authenticating all users by using their H.323-IDs and a password of qwerty2x:

```
security h323-id
security password qwerty2x
```

The next example enables security, authenticating all users by using their H.323-IDs and the password entered by the user in the H.323-ID alias he or she registers:

```
security h323-id
security password separator !
```

Now if a user registers with an H.323-ID of joe!024aqx, the gatekeeper authenticates user joe with password 024aqx, and if that is successful, registers the user with the H.323-ID of joe. If the exclamation mark is not found, the user is authenticated with the default password or a null password if no default has been configured.

The following example enables security, authenticating all users by using their E.164 IDs and the password entered by the user in the H.323-ID alias he or she registers:

```
security e164
security password separator !
```

Now if a user registers with an E.164 address of 5551212 and an H.323-ID of !hs8473q6, the gatekeeper authenticates user 5551212 and password hs8473q6. Because the H.323-ID string supplied by the user begins with the separator character, no H.323-ID is registered and the user is only known by the E.164 address.

Related Commands

Command	Description
accounting (gatekeeper)	Enables the accounting security feature on the gatekeeper.
radius-server host	Specifies a RADIUS server host.
radius-server key	Sets the authentication and encryption key for all RADIUS communications between the router and the RADIUS daemon.

sequence-numbers

To enable the generation of sequence numbers in each frame generated by the digital signal processor (DSP) for Voice over Frame Relay applications, use the **sequence-numbers** command in dial-peer configuration mode. To disable the generation of sequence numbers, use the **no** form of this command.

sequence-numbers

no sequence-numbers

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

Sequence numbers on voice packets allow the digital signal processor (DSP)s at the playout side to detect lost packets, duplicate packets or out-of-sequence packets. This helps the DSP to mask out occasional drop-outs in voice transmission at the cost of one extra byte per packet. The benefit of using sequence numbers versus the cost in bandwidth of adding an extra byte to each voice packet on the Frame Relay network must be weighed to determine whether or not to disable this function for your application.

Another factor to consider is that this command does not affect codecs that require a sequence number, such as G.726. If you are using a codec that requires a sequence number, the DSP will generate one regardless of the configuration of this command.

Examples

The following example shows how to disable the generation of sequence numbers for VoFR frames on a Cisco 2600 series or 3600 series router or on a Cisco MC3810 concentrator for VoFR dial peer 200, starting from global configuration mode:

```
dial-peer voice 200 vofr
 no sequence-numbers
```

Related Commands

Command	Description
called-number (dial-peer)	Enables an incoming VoFR call leg to get bridged to the correct POTS call leg when using a static FRF.11 trunk connection.
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.

Command	Description
cptone	Specifies a regional analog voice interface-related tone, ring, and cadence setting.
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.
session protocol (Voice over Frame Relay)	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
signal-type	Sets the signalling type to be used when connecting to a dial peer.

serial restart-delay

To set the amount of time that the router waits before trying to bring up a serial interface when it goes down, use the **serial restart-delay** command in interface configuration mode. To restore the default, use the **no** form of the command.

serial restart-delay *count*

no serial restart-delay

Syntax Description

count	Value from 0 to 900 in seconds. This is the frequency at which the hardware is reset.

Defaults

0 is the default value.

Command Modes

Interface configuration

Command History

Release	Modification
11.2 P	This command was supported.
12.0(5)XK and 12.0(7)T	Support was added for the Cisco MC3810.

Usage Guidelines

The router resets the hardware each time the serial restart timer expires. This command is often used with the dial backup feature and with the **pulse-time** command, which sets the amount of time to wait before redialing when a DTR dialed device fails to connect.

When the *count* value is set to the default of 0, the hardware is not reset when it goes down. In this way, if the interface is used to answer a call, it does not cause DTR to drop, which can cause a communications device to disconnect.

Examples

On Cisco MC3810 interface Serial 0, this examples shows the restart delay set to 0:

```
interface Serial0
 serial restart-delay 0
```

Related Commands

Command	Description
pulse-time	Enables pulsing DTR signal intervals on the serial interfaces.
show interfaces serial	Displays information about a serial interface.

session protocol

To establish a session protocol for calls between the local and remote routers via the packet network, use the **session protocol** command in dial-peer configuration mode. To reset the default value for this command, use the **no** form of this command.

session protocol *protocol*

no session protocol

Syntax Description

protocol	Specifies the call session protocol. The following session protocols are supported: **cisco**—Specifies Cisco Session Protocol session protocol. **smtp**—Specifies Simple Mail Transfer Protocol (SMTP) session protocol.

Defaults

cisco

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.

Usage Guidelines

Cisco Session Protocol (**cisco**) is the only applicable session protocol for VoIP peers. SMTP is the only applicable session protocol for Store and Forward Fax and applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

The following example selects Cisco Session Protocol as the session protocol:

```
dial-peer voice 10 voip
 session protocol cisco
```

The following example selects SMTP as the session protocol:

```
dial-peer voice 10 mmoip
 session protocol smtp
```

Related Commands

Command	Description
session target	Specifies a network-specific address for a specified dial peer.

session protocol (Voice over Frame Relay)

To establish a Voice over Frame Relay protocol for calls between the local and remote routers via the packet network, use the **session protocol** command in dial-peer configuration mode. To reset the default value for this command, use the **no** form of this command.

session protocol {**cisco-switched** | **frf11-trunk**}

no session protocol

Syntax Description

cisco-switched	Specifies proprietary Cisco VoFR session protocol. (This is the only valid session protocol for the Cisco 7200 series.)
frf11-trunk	Specifies FRF.11 session protocol.

Defaults

cisco-switched

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced for VoIP.
12.0(4)T	The **cisco-switched** and **frf11-trunk** keywords were added for VoFR dial peers.

Usage Guidelines

For Cisco-to-Cisco dial peer connections, Cisco recommends that you use the default session protocol due to the advantages it offers over a pure FRF.11 implementation. When connecting to FRF.11-compliant equipment from other vendors, use the **frf11-trunk** session protocol.

Note When using the **frf11-trunk** session protocol on Cisco 2600 series and 3600 series routers, the called-number command must also be used.

Examples

The following example shows how to configure the frf11-trunk session protocol on a Cisco 2600 series or 3600 series router for VoFR dial peer 200:

```
dial-peer voice 200 vofr
 session protocol frf11-trunk
 called-number 5552150
```

The following example shows how to configure the frf11-trunk session protocol on a Cisco MC3810 concentrator for VoFR dial peer 200:

```
dial-peer voice 200 vofr
 session protocol frf11-trunk
```

Related Commands

Command	Description
called-number (dial-peer)	Enables an incoming VoFR call leg to get bridged to the correct POTS call leg when using a static FRF.11 trunk connection.
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
cptone	Specifies a regional analog voice interface-related tone, ring, and cadence setting.
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf-relay (Voice over Frame Relay)	Enables the generation of FRF.11 Annex A frames for a dial peer.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
signal-type	Sets the signalling type to be used when connecting to a dial peer.

session target

To specify a network-specific address for a specified dial peer, use the **session target** command in dial-peer configuration mode. To restore default values for this parameter, use the **no** form of this command.

Note This command applies to all dial peers except for POTS dial peers.

Cisco 2600 series and 3600 series Voice over Frame Relay dial peers

session target *interface dlci* [*cid*]

no session target

Cisco 2600 series and 3600 series Voice over IP dial peers

session target {**ipv4:***destination-address* | **dns:**[**s.** | **d.** | **e.** | **u.**] *host-name* | **loopback:rtp** | **loopback:compressed** | **loopback:uncompressed**}

no session target

Cisco MC3810 Voice over Frame Relay dial peers

session target *interface dlci* [*cid*]

no session target

Cisco MC3810 Voice over ATM dial peers

session target *interface* **pvc** {*name* | *vpi/vci* | *vci*}

no session target

Cisco MC3810 Voice over HDLC dial peers

session target *interface*

no session target

Cisco AS5300 access servers Voice over IP dial peers

session target {**ipv4:***destination-address* | **dns:**[**s.** | **d.** | **e.** | **u.**] *host-name* | **loopback:rtp** | **loopback:compressed** | **loopback:uncompressed** | **mailto:**{*name* | **d**}@*domain-name* | **ipv4:***destination-address* | **dns:**[**s.** | **d.** | **u.** | **e.**] *host-name*}

no session target

Cisco AS5800 universal access servers Voice over IP dial peers

session target {**ipv4:***destination-address* | **dns:**[**s.** | **d.** | **e.** | **u.**] *host-name* | **loopback:rtp** | **loopback:compressed** | **loopback:uncompressed**}

no session target

Cisco 7200 series Voice over Frame Relay dial peers

session target *interface dlci*

no session target

Syntax Description

Cisco 2600 series and 3600 series Voice over Frame Relay dial peers

interface	Specifies the serial interface and interface number (slot number/port number) associated with this dial peer.
dlci	Specifies the data link connection identifier for this dial peer. The valid range is from 16 to 1007.
cid	(Optional) Specifies the DLCI subchannel to be used for data on FRF.11 calls. A CID must be specified only when the session protocol is **frf11-trunk**. When the session protocol is **cisco-switched**, the CID is dynamically allocated. The valid range is from 4 to 255. **Note** By default, CID 4 is used for data; CID 5 is used for call-control. We recommend that you select CID values between 6 and 63 for voice traffic. If the CID is greater than 63, the FRF.11 header will contain an extra byte of data.

Cisco 2600 and Cisco 3600 series Voice over IP dial peers

ipv4:*destination-address*	IP address of the dial peer.
dns:*host-name*	Indicates that the domain name server will be used to resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device. (Optional) You can use one of the following three wildcards with this keyword when defining the session target for VoIP peers: • **s.**—Indicates that the source destination pattern will be used as part of the domain name. • **d.**—Indicates that the destination number will be used as part of the domain name. • **e.**—Indicates that the digits in the called number will be reversed, periods will be added in-between each digit of the called number, and that this string will be used as part of the domain name. • **u.**—Indicates that the unmatched portion of the destination pattern (such as a defined extension number) will be used as part of the domain name.
loopback:rtp	Indicates that all voice data will be looped back to the originating source. This is applicable for VoIP peers.

loopback:compressed	Indicates that all voice data will be looped back in compressed mode to the originating source. This is applicable for POTS peers.
loopback:uncompressed	Indicates that all voice data will be looped-back in uncompressed mode to the originating source. This is applicable for POTS peers.

Cisco MC3810 Voice over Frame Relay dial peers

interface	Specifies the interface type and interface number on the Cisco MC3810. For the range of valid interface numbers for the selected interface type, enter a **?** character after the interface type.
dlci	Specifies the Frame Relay DLCI. The valid range is from 16 to 1007.
cid	(Optional) Specifies a subchannel ID for the Frame Relay DLCI. The valid range is from 4 to 255.

Cisco MC3810 Voice over ATM dial peers

interface	Specifies the interface number.
ATM *interface*	Specifies the ATM interface number on the Cisco MC3810. The only valid number is 0.
pvc	Specifies a permanent virtual circuit (pvc).
name	The PVC name.
vpi/vci	The ATM network virtual path identifier (VPI) and virtual channel identifier (VCI) of this PVC.
vci	The ATM network virtual channel identifier (VCI) of this PVC.

For the Cisco MC3810 Voice over HDLC dial peers

interface	Specifies the interface number.
serial-port-number	Specifies the serial port number on the Cisco MC3810. The valid range is 0 to 1.

Cisco AS5300 access server Voice over IP dial peers

mailto:*name*	Specific recipient e-mail address, name, or mailing list alias.
mailto	Wildcard that inserts the destination pattern of the recipient.
@*domain-name*	Specifies the appropriate domain name associated with the e-mail address.
ipv4:*destination-address*	IP address of the dial peer.

dns:*host-name*	Indicates that the domain name server will be used to resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device. (Optional) You can use one of the following three wildcards with this keyword when defining the session target for VoIP peers: • **s.**—Indicates that the source destination pattern will be used as part of the domain name. • **d.**—Indicates that the destination number will be used as part of the domain name. • **e.**—Indicates that the destination pattern is used as part of the domain name in reverse dotted format for tpc.int DNS format. For example, if the destination number is 310 555-1234 and the session target is configured as e.cisco.com, the translated DNS name will be 4.3.2.1.5.5.5.0.1.3.cisco.com. • **u.**—Indicates that the unmatched portion of the destination pattern (such as a defined extension number) will be used as part of the domain name.
loopback:rtp	Indicates that all voice data will be looped back to the originating source. This applies to VoIP peers.
loopback:compressed	Indicates that all voice data will be looped back in compressed mode to the originating source. This applies to POTS peers.
loopback:uncompressed	Indicates that all voice data will be looped back in uncompressed mode to the originating source. This applies to POTS peers.

Cisco AS5800 universal access server Voice over IP dial peers

ipv4:*destination-address*	IP address of the dial peer.
dns:*host-name*	Indicates that the domain name server will be used to resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device. (Optional) You can use one of the following three wildcards with this keyword when defining the session target for VoIP peers: • **s.**—Indicates that the source destination pattern will be used as part of the domain name. • **d.**—Indicates that the destination number will be used as part of the domain name. • **e.**—Indicates that the destination pattern is used as part of the domain name in reverse dotted format for tpc.int DNS format. For example, if the destination number is 310 555-1234 and the session target is configured as e.cisco.com, the translated DNS name will be 4.3.2.1.5.5.5.0.1.3.cisco.com. • **u.**—Indicates that the unmatched portion of the destination pattern (such as a defined extension number) will be used as part of the domain name.
loopback:rtp	Indicates that all voice data will be looped back to the originating source. This applies to VoIP peers.

loopback:compressed	Indicates that all voice data will be looped back in compressed mode to the originating source. This applies to POTS peers.
loopback:uncompressed	Indicates that all voice data will be looped back in uncompressed mode to the originating source. This applies to POTS peers.

Cisco 7200 series Voice over Frame Relay dial peers

interface	Specifies the interface type and interface number on the Cisco 7200 series router. For the range of valid interface numbers for the selected interface type, enter a **?** character after the interface type.
dlci	Specifies the Frame Relay DLCI. The valid range is from 16 to 1007.

Defaults

The default for this command is enabled with no IP address or domain name defined.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA	Support was added for VoFR, VoATM, VoHDLC, and POTS dial peers on the Cisco MC3810.
12.0(3)T	Support was added for VoIP and POTS dial peers on the Cisco AS5300.
12.0(3)XG and 12.0(4)T	Support was added for VoFR dial peers on the Cisco 2600 series and 3600 series routers. The *cid* option was added.
12.0(4)T	Support was added for VoFR and POTS dial peers on the Cisco 7200 series routers.
12.0(4)XJ	Support was added for Store and Forward Fax on the Cisco AS5300 platform.

Usage Guidelines

Use the **session target** command to specify a network-specific address or domain name for a dial peer. Whether you select a network-specific address or a domain name depends on the session protocol you select.

The **session target loopback** command is used for testing the voice transmission path of a call. The loopback point will depend on the call origination and the loopback type selected.

The **session target dns** command can be used with or without the specified wildcards. Using the optional wildcards can reduce the number of VoIP dial peer session targets you need to configure if you have groups of numbers associated with a particular router.

For VoFR dial peers, the *cid* option is not allowed when using the **cisco-switched** option for the **session protocol** command.

Use the **session target mailto** to deliver fax-mail to multiple recipients by specifying an email alias as the name argument and have that alias expanded by the mailer.

The **session target loopback** command is used for testing the voice transmission path of a call. The loopback point will depend on the call origination and the loopback type selected.

The **session target dns** command can be used with or without the specified wildcards. Using the optional wildcards can reduce the number of VoIP dial peer session targets you need to configure if you have groups of numbers associated with a particular router.

This command applies to on-ramp Store and Forward Fax functions.

Examples

The following example configures a session target using DNS for a host, voice_router, in the domain cisco.com:

```
dial-peer voice 10 voip
 session target dns:voice_router.cisco.com
```

The following example configures a session target using DNS, with the optional **u**. wildcard. In this example, the destination pattern has been configured to allow for any four-digit extension, beginning with the numbers 1310222. The optional wildcard **u.** indicates that the router will use the unmatched portion of the dialed number—in this case, the four-digit extension, to identify the dial peer. As in the previous example, the domain is cisco.com.

```
dial-peer voice 10 voip
 destination-pattern 1310222....
 session target dns:$u$.cisco.com
```

The following example configures a session target using dns, with the optional **d.** wildcard. In this example, the destination pattern has been configured for 13102221111. The optional wildcard **d.** indicates that the router will use the destination pattern to identify the dial peer in the cisco.com domain.

```
dial-peer voice 10 voip
 destination-pattern 13102221111
 session target dns:$d$.cisco.com
```

The following example configures a session target using DNS, with the optional **e.** wildcard. In this example, the destination pattern has been configured for 12345. The optional wildcard **e.** indicates that the router will reverse the digits in the destination pattern, add periods between the digits, and then use this reverse-exploded destination pattern to identify the dial peer in the cisco.com domain.

```
dial-peer voice 10 voip
 destination-pattern 12345
 session target dns:$e$.cisco.com
```

The following example configures a session target for Voice over Frame Relay on a Cisco MC3810 with a session target on serial port1 and a DLCI of 200:

```
dial-peer voice 11 vofr
 destination-pattern 13102221111
 session target serial1 200
```

The following example shows how to configure serial interface 1/0, DLCI 100 as the session target for VoFR dial peer 200 (an FRF.11 dial peer) on a Cisco 2600 series or 3600 series router, starting from global configuration mode and using the frf11-trunk session protocol:

```
dial-peer voice 200 vofr
 destination-pattern 13102221111
 called-number 5552150
 session protocol frf11-trunk
 session target serial 1/0 100 20
```

The following example configures a session target for Voice over ATM on a Cisco MC3810. The session target is sent to ATM interface 0, and for a PVC with a VCI of 20.

```
dial-peer voice 12 voatm
 destination-pattern 13102221111
 session target atm0 pvc 20
```

The following example configures a session target on serial port 0 for Voice over HDLC on a Cisco MC3810:

```
dial-peer voice 13 vohdlc
 destination-pattern 13102221111
 session target serial0
```

The following example configures a session target using dns for a host, voice_router, in the domain cisco.com:

```
dial-peer voice 10 voip
 session target dns:voice_router.cisco.com
```

The following example configures a session target using DNS, with the optional **u.** wildcard. In this example, the destination pattern has been configured to allow for any four-digit extension, beginning with the numbers 1310222. The optional wildcard **u.** indicates that the router will use the unmatched portion of the dialed number—in this case, the four-digit extension, to identify the dial peer. As in the previous example, the domain is cisco.com.

```
dial-peer voice 10 voip
 destination-pattern 1310222....
 session target dns:$u$.cisco.com
```

The following example configures a session target using DNS, with the optional **d.** wildcard. In this example, the destination pattern has been configured for 13105551111. The optional wildcard **d.** indicates that the router will use the destination pattern to identify the dial peer in the "cisco.com" domain.

```
dial-peer voice 10 voip
 destination-pattern 13105551111
 session target dns:$d$.cisco.com
```

The following example delivers fax-mail to multiple recipients:

```
dial-peer voice 10 mmoip
 session target marketing-information@mailer.example.com
```

Assuming that mailer.example.com is running sendmail, you can put the following information into its /etc/aliases file:

```
marketing-information:
 john@example.com,
 fax=+14085551212@sj-offramp.example.com
```

Related Commands

Command	Description
destination-pattern	Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer.
session protocol	Establishes a session protocol for calls between the local and remote routers through the packet network in Voice over IP.

session-timeout

To configure the lifetime of a single SSL session key, use the **session-timeout** command in settlement configuration mode. To reset to the default value of this command, use the **no** form of this command.

session-timeout *num*

no session-timeout *num*

Syntax Description

num	Defines lifetime (in seconds) of a single SSL session key.

Defaults

The default session timeout is 86,400 seconds (one day).

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

When this time limit configured by this command is exceeded, the router negotiates a new session key. Communication exchanges in progress are not interrupted when this time limit expires.

Examples

The following example shows how to configure the lifetime of a single SSL session key to one day (86,400 seconds):

```
settlement 0
 session timeout 86400
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a Settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
settlement	Enters settlement mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

settlement

To enter settlement mode and specify the attributes specific to a settlement provider, use the **settlement** command in global configuration mode. To disable the settlement provider, use the **no** form of this command.

settlement *provider-number*

no settlement *provider-number*

Syntax Description

provider-number	Specifies a digit defining a particular settlement server. The only valid entry is 0.

Defaults

The default is 0.

Command Modes

Global configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

The variable *provider-number* defines a particular Settlement provider. For Cisco IOS Release 12.1, only one clearinghouse per system is allowed, and the only valid value for *provider-number* is 0.

Examples

This example shows how to enter settlement configuration mode:

```
settlement 0
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
type	Configures an SAA-RTR operation type.

show atm video-voice address

To display the network service access point (NSAP) address for the ATM interface, enter the **show atm video-voice address** command in privileged EXEC mode.

show atm video-voice address

Syntax Description

This command has no keywords or arguments.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for the Cisco MC3810.

Usage Guidelines

Enter this command to review ATM interface NSAP addresses that have been assigned with the **atm video aesa** or **atm voice aesa** command and to ensure that ATM management is confirmed for those addresses.

Examples

On a Cisco MC3810, the following example displays ATM interface NSAP addresses:

```
router# show atm video-voice address
nsap address                                  type        ilmi status
47.0091810000000002F26D4901.00107B4832E1.FE   VOICE_AAL5  Confirmed
47.0091810000000002F26D4901.00107B4832E1.C8   VIDEO_AAL1  Confirmed
```

show bridge cable-modem

To display bridging information for a Cisco uBR900 series cable access router, enter the **show bridge cable-modem** command in privileged EXEC mode.

show bridge cable-modem *number*

Syntax Description

number	The interface number of the cable interface on the rear panel of the Cisco uBR900 series.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

Te following example is sample output for this command:

```
uBR924# show bridge cable-modem 0

Total of 300 station blocks, 298 free
Codes: P - permanent, S - self

Bridge Group 59:
```

Table 18 describes the significant fields shown in the display.

Table 18 show bridge cable-modem Field Descriptions

Field	Description
Total of 300 station blocks	Total number of forwarding database elements in the system. The memory to hold bridge entries is allocated in blocks of memory sufficient to hold 300 individual entries. When the number of free entries falls below 25, another block of memory sufficient to hold another 300 entries is allocated. Thus, the total number of forwarding elements in the system is expanded dynamically, as needed, limited by the amount of free memory in the router.
Bridge Group	The number of the bridge group to which this interface is assigned.

Related Commands

Command	Description
show dhcp	Displays the current DHCP settings on point-to-point interfaces.
show interfaces cable-modem	Displays information about the Cisco uBR900 series cable access router cable interface.

show c7200

To display the revision level information for the Cisco uBR7246 midplane, use the **show c7200** command in privileged EXEC mode.

show c7200

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 XA	This command was introduced.

Examples

The following is sample output from the **show c7200** command. The midplane EEPROM data describes the characteristics of the device's midplane chassis; the CPU EEPROM data describes the characteristics of the device CPU. The fault history buffer data provides diagnostic information used only by Cisco Customer Support Engineers.

```
router# show c7200
Network IO Interrupt Throttling:
 throttle count=0, timer count=0
 active=0, configured=0
 netint usec=3999, netint mask usec=200

UBR7200 Midplane EEPROM:
        Number of Slots          : 6
        Hardware Revision        : 1.1
        Chassis MAC Address      : 0008.cefb.fc00
        MAC Address block size   : 256
        Unknown Field (type 01B9): 2C 1F E0 00
        Unknown Field (type 01B8): 85 FF FF FF
        EEPROM format version 4
        EEPROM contents (hex):
          0x00: 04 FF 40 00 F0 01 06 41 01 01 C3 06 00 08 CE FB
          0x10: FC 00 43 01 00 C7 20 45 53 00 29 00 2E 00 3D 00
          0x20: 4C 00 34 00 36 00 87 00 81 00 83 00 86 00 84 00
          0x30: B6 00 E0 00 00 B8 DB 00 B9 2C 1F E0 00 00 B8 85
          0x40: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x50: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x60: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
          0x70: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
```

```
C7200 CPU EEPROM:
        Hardware revision 2.1            Board revision A0
        Serial number     4371856        Part number    73-1536-03
        Test history      0x0            RMA number     00-00-00
        EEPROM format version 1
        EEPROM contents (hex):
          0x20: 01 15 02 01 00 42 B5 90 49 06 00 03 00 00 00 00
          0x30: 50 00 00 00 FF FF FF FF FF FF FF FF FF FF FF FF

Fault History Buffer:
7200 Software (UBR7200-P-M), Experimental Version 11.3(19980514:205205)
[xx-xx_2 232]
Compiled Fri 12-Jun-98 19:20 by johnchen
Signal = 23, Code = 0x24, Uptime 00:02:09
$0 : 00000000, AT : 00000000, v0 : 00000000, v1 : 00000004
a0 : 00000000, a1 : 0000FF00, a2 : 00000006, a3 : 00000002
t0 : 00000020, t1 : 3401FF01, t2 : 3401C100, t3 : FFFF00FF
t4 : 6027E180, t5 : 30443044, t6 : 30384330, t7 : 30783630
s0 : 00000000, s1 : 608BFD88, s2 : 606D9E4C, s3 : 60B43E0C
s4 : 608BFD88, s5 : 0000004A, s6 : 00000000, s7 : 608BFF9C
t8 : 00009BCB, t9 : 00000000, k0 : 3041D001, k1 : BF800000
gp : 6083B400, sp : 60BC4CA0, s8 : 608BFDF8, ra : 602797EC
EP6027AE58, SREG : 3401FF03, Cause : 00000424
```

show cable flap-list

To display the cable flap-list on a Cisco uBR7200 series cable router, use the **show cable flap-list** command in privileged EXEC mode.

show cable flap-list [**sort-flap** | **sort-time**]

Syntax Description

sort-flap	(Optional) Sort by number of times the cable modem has flapped.
sort-time	(Optional) Sort most recent time the cable modem is detected to have flapped.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following examples show the return for flap-list tables sorted by MAC address and by time:

```
CMTS01# show cable flap-list sort-flap
Mac Addr        CableIF     Ins    Hit   Miss    CRC  P-Adj   Flap    Time
.1eab.2c0b      C6/0 U0     108    318     27      0      0    108 Sep 10 15:26:56
.1eb2.bb07      C6/0 U0       0    293     31      1      1      1 Sep 10 15:15:49
.7b6b.71cd      C6/0 U0       1    288     32      0      0      1 Sep 10 15:12:13
.1eb2.bb8f      C6/0 U0       1    295     30      0      0      1 Sep 10 15:11:44
router#

CMTS01# show cable flap-list sort-time
Mac Addr        CableIF     Ins    Hit   Miss    CRC  P-Adj   Flap    Time
00e0.2222.2202 C4/0 U0      464   2069    242      0    421    885 Oct 16 22:47:23
0010.7b6b.57e1 C4/0 U0        0   2475     43      0   1041   1041 Oct 16 22:47:04
```

Table 19 describes the fields displayed by the **show flap-list** command.

Table 19 show cable flap-list Command Field Descriptions

Field	Description
Mac Addr	The customer account or street address.
CableIF	The physical port, including the upstream port.
Ins	The number of times the modem comes up and inserts itself into the network. It can indicate intermittent downstream sync loss or DHCP or modem registration problems.

Table 19 show cable flap-list Command Field Descriptions (continued)

Field	Description
Hit	The number of times the modem responds to MAC layer keepalive messages. (The minimum hit rate is once per 30 seconds. It can indicate intermittent upstream, laser clipping, or common-path distortion.
Miss	The number of times the modem misses the MAC layer keep-alive message. An 8% miss rate is normal for the Cisco cable modem cards. It can indicate intermittent upstream, laser clipping, or common-path distortion.
CRC	The number of Cyclic Redundancy Check errors from this modem. It can indicate intermittent upstream, laser clipping, or common-path distortion.
P-Adj	The number of times the headend instructed the modem to adjust transmit (TX) power more than 3 dB. It can indicate amplifier degradation, poor connections, or thermal sensitivity.
Flap	The sum of P-Adj and Ins values. Modems with high flap counts will have high SIDs and might not register.
Time	The most recent time that the modem dropped the connection.

show cable hop

To display cable-hop statistics on a Cisco uBR7200 series cable router, use the **show cable hop** command in EXEC mode.

show cable hop [*cable-if*] [**upstream** *portnum*]

Syntax Description

cable-if	(Optional) Specifies the cable interface.
upstream *portnum*	(Optional) Specifies the upstream port for which you wish to display the frequency hop status.

Defaults

No default behavior or values.

Command Modes

EXEC

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following examples show output from the **show cable hop** and **show cable hop upstream** commands.

```
ubr7200> show cable hop
Upstream     Port        Poll Missed Min     Missed Hop   Hop     Corr     Uncorr
Port         Status      Rate Poll   Poll    Poll   Thres Period  FEC      FEC
                         (ms) Count  Sample Pcnt   Pcnt  (sec)   Errors   Errors
Cable4/0/U0 down        1000 * * *   frequency not set    * * * 0        0
Cable4/0/U1 down        1000 * * *   frequency not set    * * * 0        0
Cable4/0/U2 down        1000 * * *   frequency not set    * * * 0        0
Cable4/0/U3 down        1000 * * *   frequency not set    * * * 0        0
Cable4/0/U4 down        1000 * * *   frequency not set    * * * 0        0
Cable4/0/U5 down        1000 * * *   frequency not set    * * * 0        0
Cable5/0/U0 down        1000 * * *   interface is down    * * * 0        0
Cable5/0/U1 down        1000 * * *   frequency not set    * * * 0        0
Cable5/0/U2 down        1000 * * *   interface is down    * * * 0        0
Cable5/0/U3 down        1000 * * *   interface is down    * * * 0        0
Cable5/0/U4 down        1000 * * *   interface is down    * * * 0        0
Cable5/0/U5 down        1000 * * *   interface is down    * * * 0        0
Cable6/0/U0 down        1000 * * *   interface is down    * * * 0        0

CMTS-ubr7223#show cable hop c2/0 upstream 2
Upstream     Port        Poll Missed Min     Missed Hop   Hop     Corr     Uncorr
Port         Status      Rate Poll   Poll    Poll   Thres Period  FEC      FEC
                         (ms) Count  Sample Pcnt   Pcnt  (sec)   Errors   Errors
Cable2/0/U2 admindown   1000 * * *   frequency not set    * * * 0        0
```

Table 20 describes the fields shown in the **show cable hop** and **show cap hop upstream** examples.

Table 20 show cable hop Command Field Descriptions

Field	Description
Upstream Port	The upstream port for this information line.
Port Status	Lists the status of the port. Valid states are down if frequency is unassigned or admindown if the port is shut down. If the port is up, this column shows the center frequency of the channel.
Poll Rate	The rate that station maintenance polls are generated (in milliseconds).
Missed Poll Count	The number of missing polls.
Min Poll Sample	The number of polls in the sample.
Missed PollPcnt	The ratio of missing polls to the number of polls, expressed as a percentage.
Hop Thres Pcnt	The level that the missed poll percentage must exceed to trigger a frequency hop, expressed as a percentage.
Hop Period	The maximum rate that frequency hopping will occur (in seconds).
Corr FEC Errors	The number of correctable (forward error corrections) FEC errors on this upstream port. FECs measure noise.
Uncorr FEC Errors	The number of uncorrectable FEC errors on this upstream port.

Related Commands

Command	Description
show cable modem	Displays cable modem configuration settings.
show cable host	Displays the statistics for the host behind the cable modem.

show cable modem

To view configuration settings on the Cisco uBR7200 series cable router, use the **show cable** command in privileged EXEC mode.

show cable modem [*ip-address* | *mac-address*]

Syntax Description

ip-address	(Optional) Specify the IP address of the cable modem.
mac-address	(Optional) Specify the MAC address of the cable modem.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 XA	This command was introduced.
12.0(3)T	The *mac-address* keyword was added and the output was modified.
12.0(4)XI	The output for this command was modified to identify the primary SID.

Usage Guidelines

This command displays information on all cable modems on the network, or on the particular cable modem you specify.

Examples

The following is sample output from the **show cable modem** command:

```
ubr7200# show cable modem
Interface    Prim Online     Timing Rec     QoS CPE IP address      MAC address
             Sid  State      Offset Power
Cable2/0/U0 1    online      2288    0.50  4   0    172.16.30.66    0010.7bb3.fb45
Cable2/0/U0 2    online      2288    0.50  4   0    172.16.30.68    0010.7bb3.fb7b
Cable2/0/U0 3    init(i)     2280    0.00  2   0    172.16.30.69    0010.9500.05e
```

Table 21 describes the fields shown in the **show cable modem** example.

Table 21 show cable modem Field Descriptions

Field	Description
Interface	The interface on which the cable modem has an active connection.
Prim Sid	The primary service identifier assigned to the modem.
Online State	The status of the modem.
Timing Offset	The cable modem current timing adjustments.
Rec Power	The nominal receive power in decibels for this SID.

Table 21 show cable modem Field Descriptions (continued)

Field	Description
QoS	The service cass assigned to the modem.
CPE	The number of CPE devices (PCs, Macintoshes, UNIX workstations, and so on.) behind this cable modem.
IP address	The IP address of the modem.
MAC address	The media access layer address of the modem.

Related Commands

Command	Description
show cable burst-profile	Displays the upstream data burst profiles used to configure the upstream PHY.
show cable modulation-profile	Displays modulation profile group information.
show interface cable sid	Displays cable interface information.

show cable modulation-profile

To display modulation profile group information for a Cisco uBR7200 series cable router, use the **show cable modulation-profile** command in privileged EXEC mode.

show cable modulation-profile [*profile*] [*iuc-code*]

Syntax Description

profile	(Optional) Profile number. Valid values are from 1 to 8.
iuc-code	(Optional) Internal usage code. Valid options are: **initial**—Initial Ranging Burst **long**—Long Grant Burst **request**—Request Burst **short**—Short Grant Burst **station**—Station Ranging Burst

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 XA	This command was introduced.

Usage Guidelines

This command displays modulation profile group information. A modulation profile is a collection of six burst profiles that are sent out in an upstream channel descriptors (UCD) message to configure a cable modem transmit parameters for the following upstream message types: request, initial maintenance, station maintenance, short grant, and long grant.

Examples

The following is sample output from the **show cable modulation-profile** command:

```
CMTS01# show cable modulation-profile 1
Mo IUC      Type  Preamb Diff FEC     FEC   Scrambl Max  Guard Last Scrambl Preamb
                  length enco T       CW    seed    B    time  CW           offset
                              bytes      size            size  size  short
1  request  qpsk  64     no   0x0     0x10  0x152   1    8     no   yes     56
1  initial  qpsk  128    no   0x5     0x22  0x152   0    48    no   yes     0
1  station  qpsk  128    no   0x5     0x22  0x152   0    48    no   yes     0
1  short    qpsk  72     no   0x5     0x4B  0x152   0    8     no   yes     48
```

Table 22 describes the fields shown in the **show cable modulation-profile** display.

Table 22 show cable modulation-profile Field Descriptions

Field	Description
Mo	Modulation profile group number. A modulation profile group is the set of burst profiles that define upstream transmit characteristics for the various types of upstream transmission classes.
IUC	Interval usage code. Each upstream transmit burst belongs to a class which is given a number called the IUC. Bandwidth maps messages (MAP) by IUC codes used to allocate upstream time slots. The following types are currently defined: • Request—bandwidth request slot • Initial Maintenance—initial link registration contention slot • Station Maintenance—link keep-alive slot • Short Data Grant—short data burst slot • Long Data Grant—long data burst slot
Type	Modulation type.
Preamb length	Preamble length.
Diff enco	Differential encoding enabled (yes) or not enabled (no).
FEC T bytes	Number of bytes that can be corrected for each FEC code word.
FEC CW size	Size, in bytes, of the FEC code word.
Scrambl seed	Scrambler seed value in hex format.
Max B size	Maximum burst size.
Guard time size	Time between successive bursts measured in symbols.
Last CW short	Handling of FEC for shortened last code word.
Scrambl	Scrambler enabled (yes) or not enabled (no).
Preamb offset	The bits to be used for the preamble value.

Related Commands

Command	Description
show cable burst-profile	Displays the upstream data burst profiles used to configure the upstream PHY.
show cable hop	Displays cable modem configuration settings.
show interface cable sid	Displays cable interface information.

show cable noise

To display cable-noise statistics on a Cisco uBR7200 series cable router, use the **show cable noise** command in EXEC mode.

show cable *slot/port* **noise**

Syntax Description

slot/port	Specifies the slot and port number for which information is to be displayed.

Defaults

No default behavior or values.

Command Modes

EXEC

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Examples

The following example shows how to display cable modem noise statistics:

```
ubr7223# show cable 6/0 noise
```

Related Commands

Command	Description
show cable modem	Displays cable modem configuration settings.

show cable qos permission

To display the status of permissions for changing quality of service tables on a Cisco uBR7200 series cable router, use the **show cable qos permission** command in privileged EXEC mode.

cable qos permission

Syntax Description

This command has no keywords or arguments.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following example displays the output of the **show cable qos permission** command:

```
CMTS01# show cable qos permission

Create by SNMP  Update by SNMP  Create by modems
yes             yes             yes
```

Table 23 describes the fields shown in the **show cable qos permission** displays.

Table 23 show cable qos permission Command Field Descriptions

Field	Description
Create by SNMP	Indicates permission setting for creation of QoS table entries by Simple Network Management Protocol (SNMP).
Update by SNMP	Indicates permission setting for creation of QoS table entries by modem registration requests.
Create by modems	Indicates permission setting for dynamic updating of QoS table entries by Simple Network Management Protocol (SNMP).

Related Commands

Command	Description
cable qos permission	Specifies permission for updating the cable router QoS table.
cable qos profile	Configures a QoS profiles.
show cable qos profile	Displays cable router QoS profiles.

show cable qos profile

To display quality of service profiles for a Cisco uBR7200 series cable router, use the **show cable qos profile** command in privileged EXEC mode.

show cable qos profile *service class*

Syntax Description

service class	Displays cable QoS table.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following example displays the QoS tables for profiles 1, 2, 3, and 4:

```
CMTS01# show cable qos profile
Service Prio Max       Guarantee Max        Max tx TOS  TOS   Create     B
class        upstream  upstream  downstream burst  mask value by         priv
             bandwidth bandwidth bandwidth                               enab
1       0    0         0         0          0      0x0  0x0   cmts       no
2       0    64000     0         1000000    0      0x0  0x0   cmts       no
3       0    1000      0         1000       0      0x0  0x0   cmts       no
4       7    2000000   100000    4000000    0      0x0  0x0   cm         yes
```

Table 24 describes the fields shown in the **show cable qos profile** displays.

Table 24 show cable qos profile Command Field Descriptions

Field	Description
Service Class	Profile number.
Prio	Priority level.
Max upstream bandwidth	Maximum upstream bandwidth.
Guarantee upstream bandwidth	Guaranteed minimum upstream bandwidth.
Max downstream bandwidth	Maximum downstream bandwidth.
Max tx burst	Maximum transmit burst size in minislots.
Tos mask	Hex value of the mask bits.
Tos value	Value of the mask byte.

Table 24 show cable qos profile Command Field Descriptions (continued)

Field	Description
Create by	Identity of the profile creator.
B priv enab	Reports yes if Baseline Privacy is enabled for this QoS profile. Reports no if Baseline Privacy is not enabled for this Qos profile.

Related Commands

Command	Description
cable qos permission	Configures permissions for updating the QoS table.
cable qos profile	Displays QoS profiles.
show cable noise	Displays the status of permissions for changing QoS tables.

show cable spectrum-group

To display information about spectrum groups on a Cisco uBR7200 series cable router, use the **show cable spectrum-group** command in privileged EXEC mode.

show cable spectrum-group [*groupnum*]

Syntax Description

groupnum	(Optional) Displays information about the specified group number. If no group number is specified, information for all spectrum groups is displayed.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following is sample output from the **show cable spectrum-group** command for the upstream spectrum group named **sales**:

```
CMTS01# show cable spectrum-group sales
Spectrum  Frequency Band   Upstream   Time          Time            Input     Shared
 Group    (MHz)            Port       Available     Delete        PowerLevel Topology
   4      5.000-40.000                                                 5        N
   4      5.000                                                        5        N
   4      5.000-40.000               Mon 12:00:00  Mon 12:00:00        5        N
   4      5.000                      Mon 12:00:00                      5        N
```

Table 25 describes the fields shown in the **show cable spectrum-group** displays.

Table 25 show cable spectrum-group Command Field Descriptions

Field	Description
Spectrum-Group	Identifies the spectrum group.
Frequency Band (MHz)	Identifies the upper and lower ranges of the frequency for this spectrum group.
Upstream Port	Identifies the upstream port number.
Time Available	Identifies the day and time of day when this group is available.
Time Delete	Identifies the day and time of day when this group will be deleted.

Table 25 show cable spectrum-group Command Field Descriptions (continued)

Field	Description
Input PowerLevel	Identifies the assigned decibels per millivolt (dBmV) input level.
Shared Topology	Indicates if upstreams are physically combined (share the same combiner group). Y or yes values indicate that upstreams which are members of the spectrum group are combined and cannot be assigned overlapping frequency bands. N or no values indicate that upstreams which are members of the spectrum group are not combined and can be assigned overlapping frequency bands.

Related Commands

Command	Description
show cable burst-profile	Displays the upstream data burst profiles used to configure the upstream PHY.
show cable hop	Displays cable modem configuration settings.
show cable modulation-profile	Displays modulation profile group information.

show call active

To show active call information for a voice call or fax transmission in progress, use the **show call active** command in privileged EXEC mode.

show call active {**voice** | **fax**}

Syntax Description

voice	Specifies that the active call table displays voice call information.
fax	Specifies that the active call table displays fax call information.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.

Usage Guidelines

Use the **show call active** privileged EXEC command to display the contents of the active call table. If you use the **voice** keyword, the active call table displays information about all of the voice calls currently connected through the router or access server. If you use the **fax** keyword, the active call table shows all of the fax calls currently connected through the router.

This command applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

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

```
router# show call active voice
 GENERIC:
SetupTime=179388054 ms
Index=1
PeerAddress=+5....
PeerSubAddress=
PeerId=5
PeerIfIndex=32
LogicalIfIndex=29
ConnectTime=179389793 ms
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=2
TransmitPackets=532
TransmitBytes=10640
ReceivePackets=147
ReceiveBytes=2940

TELE:
```

```
ConnectionId=[0xE3EA3FF8 0xFF6D0105 0x0 0x6AEC71E4]
TxDuration=23230 ms
VoiceTxDuration=2940 ms
FaxTxDuration=0 ms
CoderTypeRate=g729r8
NoiseLevel=-84
ACOMLevel=20
OutSignalLevel=-66
InSignalLevel=-66
InfoActivity=2
ERLLevel=20
SessionTarget=

GENERIC:
SetupTime=179388237 ms
Index=1
PeerAddress=+3622
PeerSubAddress=
PeerId=3
PeerIfIndex=31
LogicalIfIndex=0
ConnectTime=179389793 ms
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=143
TransmitBytes=2860
ReceivePackets=580
ReceiveBytes=11600

VOIP:
ConnectionId[0xE3EA3FF8 0xFF6D0105 0x0 0x6AEC71E4]
RemoteIPAddress=172.24.96.200
RemoteUDPPort=16422
RoundTripDelay=37 ms
SelectedQoS=best-effort
SessionProtocol=cisco
SessionTarget=ipv4:172.24.96.200
OnTimeRvPlayout=9920
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=30 ms
ReceiveDelay=30 ms
VAD = enabled
CoderTypeRate=g729r8
```

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

```
router# show call active fax brief

<ID>: <start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state> \
  tx:<packets>/<bytes> rx:<packets>/<bytes> <state>
IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
Tele <int>: tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l> dBm

1     : 22021hs.1 +2263 pid:0 Answer wook song active
```

```
tx:0/0 rx:0/41190
IP 0.0.0.0 AcceptedMime:2 DiscardedMime:1

1    : 23193hs.1 +1091 pid:3469 Originate 527.... active
tx:10/13838 rx:0/0
Tele : tx:31200/10910/20290ms  noise:-1 acom:-1  i/0:0/0 dBm
```

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

```
router# show call active fax

GENERIC:
SetupTime=22021 ms
Index=1
PeerAddress=wook song
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=24284
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=10
TransmitPackets=0
TransmitBytes=0
ReceivePackets=0
ReceiveBytes=41190

MMOIP:
ConnectionId[0x37EC7F41 0xB0110001 0x0 0x35C34]
RemoteIPAddress=0.0.0.0
SessionProtocol=SMTP
SessionTarget=
MessageId=
AccountId=
ImgEncodingType=MH
ImgResolution=fine
AcceptedMimeTypes=2
DiscardedMimeTypes=1
Notification=None

GENERIC:
SetupTime=23193 ms
Index=1
PeerAddress=527....
PeerSubAddress=
PeerId=3469
PeerIfIndex=157
LogicalIfIndex=30
ConnectTime=24284
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=10
TransmitPackets=5
TransmitBytes=6513
ReceivePackets=0
ReceiveBytes=0

TELE:
ConnectionId=[0x37EC7F41 0xB0110001 0x0 0x35C34]
TxDuration=24010 ms
```

```
FaxTxDuration=10910 ms
FaxRate=14400
NoiseLevel=-1
ACOMLevel=-1
OutSignalLevel=0
InSignalLevel=0
InfoActivity=0
ERLLevel=-1
SessionTarget=
ImgPages=0
```

Table 26 provides an alphabetical listing of the **show call active** command fields and a description of each field.

Table 26 show call active Command Field Descriptions

Field	Description
ACOM Level	Current ACOM level for this call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for this call.
CallOrigin	Call origin: answer or originate.
CallState	Current state of the call.
CoderTypeRate	Negotiated coder transmit rate of voice/fax compression during this call.
ConnectionId	Global call identifier for this gateway call.
ConnectTime	Time at which the call was connected.
Dial-Peer	Tag of the dial peer sending this call.
ERLLevel	Current Echo Return Loss (ERL) level for this call.
FaxTxDuration	Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received in time from voice gateway for this call.
GapFillWith Redundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because voice data was lost or not received in time from voice gateway for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call. Examples of such pullout are frame-eraser or frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWith Silence	Duration of voice signal replaced with silence because voice data was lost or not received in time for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during this call.
Index	Dial-peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call.

Table 26 show call active Command Field Descriptions (continued)

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

Related Commands

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

Command	Description
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 call application voice

To define the names of the audio files the interactive voice response (IVR) script will play, the operation of the abort keys, which prompts are used, and caller interaction, the **show call application voice** command in EXEC mode.

show call application voice [*name* | **summary**]

Syntax Description

name	(Optional) The name of the desired IVR application.
summary	(Optional) Enter this field to display a one line summary. If the command is entered without summary, a complete detailed description is displayed of the application.

Defaults

No default behavior or values.

Command Modes

EXEC

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Usage Guidelines

If the name of a specific application is entered, it will give information about that application.

If the **summary** keyword is entered a one line summary will be displayed about each application.

If the command is entered without the **summary**, a detailed description of the entered IVR application is displayed.

Examples

This example shows the output for the clid_authen_collect IVR script:

```
sblab115> show call application voice clid_authen_collect
Application clid_authen_collect has 10 states with 0 calls active
 State start has 1 actions and 5 events
    Do Action IVR_ACT_AUTHENTICATE. accountName=ani, pinName=dnis
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_CALL_SETUP_IND do action IVR_ACT_CALL_SETUP_ACK
          and goto state start
    If Event IVR_EV_AAA_SUCCESS goto state collect_dest
    If Event IVR_EV_AAA_FAIL goto state get_account
 State end has 1 actions and 3 events
    Do Action IVR_ACT_END.
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_CALL_DISCONNECT_DONE do action IVR_ACT_CALL_DESTROY
          and do nothing
```

```
State get_account has 4 actions and 7 events
    Do Action IVR_ACT_PLAY.
            URL: flash:enter_account.au
            allowInt=1, pContent=0x60E4C564
    Do Action IVR_ACT_ABORT_KEY. abortKey=*
    Do Action IVR_ACT_TERMINATION_KEY. terminationKey=#
    Do Action IVR_ACT_COLLECT_PATTERN. Pattern account is .+
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_PAT_COL_SUCCESS goto state get_pin
            patName=account
    If Event IVR_EV_ABORT goto state get_account
    If Event IVR_EV_PLAY_COMPLETE do nothing
    If Event IVR_EV_TIMEOUT goto state get_account count=0
    If Event IVR_EV_PAT_COL_FAIL goto state get_account
 State get_pin has 4 actions and 7 events
    Do Action IVR_ACT_PLAY.
            URL: flash:enter_pin.au
            allowInt=1, pContent=0x0
    Do Action IVR_ACT_ABORT_KEY. abortKey=*
    Do Action IVR_ACT_TERMINATION_KEY. terminationKey=#
    Do Action IVR_ACT_COLLECT_PATTERN. Pattern pin is .+
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_PAT_COL_SUCCESS goto state authenticate
            patName=pin
    If Event IVR_EV_PLAY_COMPLETE do nothing
    If Event IVR_EV_ABORT goto state get_account
    If Event IVR_EV_TIMEOUT goto state get_pin count=0
    If Event IVR_EV_PAT_COL_FAIL goto state get_pin
 State authenticate has 1 actions and 5 events
    Do Action IVR_ACT_AUTHENTICATE. accountName=account, pinName=pin
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_AAA_SUCCESS goto state collect_dest
    If Event IVR_EV_TIMEOUT do nothing count=0
    If Event IVR_EV_AAA_FAIL goto state authenticate_fail
 State collect_dest has 4 actions and 8 events
    Do Action IVR_ACT_PLAY.
            URL: flash:enter_destination.au
            allowInt=1, pContent=0x0
    Do Action IVR_ACT_ABORT_KEY. abortKey=*
    Do Action IVR_ACT_TERMINATION_KEY. terminationKey=#
    Do Action IVR_ACT_COLLECT_DIALPLAN.
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_PLAY_COMPLETE do nothing
    If Event IVR_EV_ABORT goto state collect_dest
    If Event IVR_EV_TIMEOUT goto state collect_dest count=0
    If Event IVR_EV_DIAL_COL_SUCCESS goto state place_call
    If Event IVR_EV_DIAL_COL_FAIL goto state collect_dest
    If Event IVR_EV_TIMEOUT goto state collect_dest count=0
 State place_call has 1 actions and 4 events
    Do Action IVR_ACT_PLACE_CALL.
            destination= called=
            calling=      account=
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_CALL_UP goto state active
    If Event IVR_EV_CALL_FAIL goto state place_fail
 State active has 0 actions and 2 events
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
```

```
 State authenticate_fail has 1 actions and 2 events
    Do Action IVR_ACT_PLAY.
            URL: flash:auth_failed.au
            allowInt=0, pContent=0x0
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
 State place_fail has 1 actions and 2 events
    Do Action IVR_ACT_PLAY_FAILURE_TONE.
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing

sblab115> show call application voice clid_authen_collect
Application clid_authen_collect has 10 states with 0 calls active
 State start has 1 actions and 5 events
    Do Action IVR_ACT_AUTHENTICATE. accountName=ani, pinName=dnis
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_CALL_SETUP_IND do action IVR_ACT_CALL_SETUP_ACK
          and goto state start
    If Event IVR_EV_AAA_SUCCESS goto state collect_dest
    If Event IVR_EV_AAA_FAIL goto state get_account
 State end has 1 actions and 3 events
    Do Action IVR_ACT_END.
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_CALL_DISCONNECT_DONE do action IVR_ACT_CALL_DESTROY
          and do nothing
 State get_account has 4 actions and 7 events
    Do Action IVR_ACT_PLAY.
            URL: flash:enter_account.au
            allowInt=1, pContent=0x60E4C564
    Do Action IVR_ACT_ABORT_KEY. abortKey=*
    Do Action IVR_ACT_TERMINATION_KEY. terminationKey=#
    Do Action IVR_ACT_COLLECT_PATTERN. Pattern account is .+
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_PAT_COL_SUCCESS goto state get_pin
            patName=account
    If Event IVR_EV_ABORT goto state get_account
    If Event IVR_EV_PLAY_COMPLETE do nothing
    If Event IVR_EV_TIMEOUT goto state get_account count=0
    If Event IVR_EV_PAT_COL_FAIL goto state get_account
 State get_pin has 4 actions and 7 events
    Do Action IVR_ACT_PLAY.
            URL: flash:enter_pin.au
            allowInt=1, pContent=0x0
    Do Action IVR_ACT_ABORT_KEY. abortKey=*
    Do Action IVR_ACT_TERMINATION_KEY. terminationKey=#
    Do Action IVR_ACT_COLLECT_PATTERN. Pattern pin is .+
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_PAT_COL_SUCCESS goto state authenticate
            patName=pin
    If Event IVR_EV_PLAY_COMPLETE do nothing
    If Event IVR_EV_ABORT goto state get_account
    If Event IVR_EV_TIMEOUT goto state get_pin count=0
    If Event IVR_EV_PAT_COL_FAIL goto state get_pin
 State authenticate has 1 actions and 5 events
    Do Action IVR_ACT_AUTHENTICATE. accountName=account, pinName=pin
    If Event IVR_EV_DEFAULT goto state end
    If Event IVR_EV_CALL_DIGIT do nothing
    If Event IVR_EV_AAA_SUCCESS goto state collect_dest
    If Event IVR_EV_TIMEOUT do nothing count=0
    If Event IVR_EV_AAA_FAIL goto state authenticate_fail
```

```
State collect_dest has 4 actions and 8 events
   Do Action IVR_ACT_PLAY.
           URL: flash:enter_destination.au
           allowInt=1, pContent=0x0
   Do Action IVR_ACT_ABORT_KEY. abortKey=*
   Do Action IVR_ACT_TERMINATION_KEY. terminationKey=#
   Do Action IVR_ACT_COLLECT_DIALPLAN.
   If Event IVR_EV_DEFAULT goto state end
   If Event IVR_EV_CALL_DIGIT do nothing
   If Event IVR_EV_PLAY_COMPLETE do nothing
   If Event IVR_EV_ABORT goto state collect_dest
   If Event IVR_EV_TIMEOUT goto state collect_dest count=0
   If Event IVR_EV_DIAL_COL_SUCCESS goto state place_call
   If Event IVR_EV_DIAL_COL_FAIL goto state collect_dest
   If Event IVR_EV_TIMEOUT goto state collect_dest count=0
State place_call has 1 actions and 4 events
   Do Action IVR_ACT_PLACE_CALL.
           destination= called=
           calling=      account=
   If Event IVR_EV_DEFAULT goto state end
   If Event IVR_EV_CALL_DIGIT do nothing
   If Event IVR_EV_CALL_UP goto state active
   If Event IVR_EV_CALL_FAIL goto state place_fail
State active has 0 actions and 2 events
   If Event IVR_EV_DEFAULT goto state end
   If Event IVR_EV_CALL_DIGIT do nothing
State authenticate_fail has 1 actions and 2 events
   Do Action IVR_ACT_PLAY.
           URL: flash:auth_failed.au
           allowInt=0, pContent=0x0
   If Event IVR_EV_DEFAULT goto state end
   If Event IVR_EV_CALL_DIGIT do nothing
State place_fail has 1 actions and 2 events
   Do Action IVR_ACT_PLAY_FAILURE_TONE.
   If Event IVR_EV_DEFAULT goto state end
   If Event IVR_EV_CALL_DIGIT do nothing
```

Related Commands

Command	Description
call application voice	Defines the name to be used for an application and indicates the location of the appropriate IVR script to be used with this application.
call application voice load	Reloads the designated TCL script.

show call history

To display the fax call history table for a fax transmission, use the **show call history** command in privileged EXEC mode.

show call history {**voice** | **fax**} [**last** *number* | **brief**]

Syntax Description

voice	Specifies that the call history tables displays voice call information.
fax	Specifies that the call history table displays fax call information.
last *number*	(Optional) Displays the last calls connected, where the number of calls displayed is defined by the argument *number*. Valid values are from 1 to 2147483647.
brief	(Optional) Displays a truncated version of the call history table.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.

Usage Guidelines

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

Use the **show call history fax** command to display the fax call history table. The fax call history table contains a listing of all fax calls connected through this router in descending time order since Store and Forward Fax was enabled. You can display subsets of the fax call history table by using the **show call history** command with specific keywords. To display the last calls connected through this router, use the keyword **last**, and define the number of calls to be displayed with the argument *number*. To display a truncated version of the call history table, use the **brief** keyword. This command applies to both on-ramp and off-ramp Store and Forward Fax functions.

Examples

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

```
router# show call history voice brief
<ID>: <start>hs.<index> +<connect> +<disc> pid:<peer_id> <direction> <addr>
 tx:<packets>/<bytes> rx:<packets>/<bytes> <disc-cause>(<text>)
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms delay:<last>/<min>/<max>ms <codec>
 Telephony <int>: tx:<tot>/<voice>/<fax>ms <codec> noise:<lvl>dBm acom:<lvl>dBm
```

```
234   : 158305740hs.1280 +241 +9199 pid:0 Answer +3...
 tx:3804/76080 rx:1358/27160 10  (normal call clearing.)
 IP 172.24.96.200:16468 rtt:33ms pl:25990/0ms delay:30/30/70ms g729r8

234   : 158305745hs.1281 +236 +9195 pid:6 Originate +68888
 tx:1358/27160 rx:3804/76080 10  (normal call clearing.)
 Telephony 0:D:22: tx:91850/76080/0ms g729r8 noise:-84dBm acom:20dBm

235   : 158344850hs.1282 +230 +28773 pid:0 Answer +3...
 tx:11063/221260 rx:4604/92080 10  (normal call clearing.)
 IP 172.24.96.200:16474 rtt:41ms pl:88260/290ms delay:40/30/130ms g729r8

235   : 158344856hs.1283 +224 +28769 pid:6 Originate +68888
 tx:4604/92080 rx:11063/221260 10  (normal call clearing.)
 Telephony 0:D:22: tx:287590/221280/0ms g729r8 noise:-75dBm acom:20dBm
```

The following is sample output from the **show call history fax brief** command:

```
router# show call history fax brief
<ID>: <start>hs.<index> +<connect> +<disc> pid:<peer_id> <direction> <addr>
 tx:<packets>/<bytes> rx:<packets>/<bytes> <disc-cause>(<text>)
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
 Telephony <int>: tx:<tot>/<voice>/<fax>ms <codec> noise:<lvl>dBm acom:<lvl>dBm

2     : 5996450hs.25 +-1 +3802 pid:100 Answer 408
 tx:0/0 rx:0/0 1F  (T30 T1 EOM timeout)
 Telephony : tx:38020/38020/0ms g729r8 noise:0dBm acom:0dBm

2     : 5996752hs.26 +-1 +3500 pid:110 Originate uut1@linux2.allegro.com
 tx:0/0 rx:0/0 3F  (The e-mail was not sent correctly. Remote SMTP server said: 354 )
 IP 14.0.0.1 AcceptedMime:0 DiscardedMime:0

3     : 6447851hs.27 +1111 +3616 pid:310 Originate 576341.
 tx:11/14419 rx:0/0 10  (Normal connection)
 Telephony : tx:36160/11110/25050ms g729r8 noise:115dBm acom:-14dBm

3     : 6447780hs.28 +1182 +4516 pid:0 Answer
 tx:0/0 rx:0/0 10  (normal call clearing.)
 IP 0.0.0.0 AcceptedMime:0 DiscardedMime:0

4     : 6464816hs.29 +1050 +3555 pid:310 Originate 576341.
 tx:11/14413 rx:0/0 10  (Normal connection)
 Telephony : tx:35550/10500/25050ms g729r8 noise:115dBm acom:-14dBm

4     : 6464748hs.30 +1118 +4517 pid:0 Answer
 tx:0/0 rx:0/0 10  (normal call clearing.)
 IP 0.0.0.0 AcceptedMime:0 DiscardedMime:0

5     : 6507900hs.31 +1158 +2392 pid:100 Answer 4085763413
 tx:0/0 rx:3/3224 10  (Normal connection)
 Telephony : tx:23920/11580/12340ms g729r8 noise:0dBm acom:0dBm

5     : 6508152hs.32 +1727 +2140 pid:110 Originate uut1@linux2.allegro.com
 tx:0/2754 rx:0/0 3F  (service or option not available, unspecified)
 IP 14.0.0.4 AcceptedMime:0 DiscardedMime:0

6     : 6517176hs.33 +1079 +3571 pid:310 Originate 576341.
 tx:11/14447 rx:0/0 10  (Normal connection)
 Telephony : tx:35710/10790/24920ms g729r8 noise:115dBm acom:-14dBm

6     : 6517106hs.34 +1149 +4517 pid:0 Answer
 tx:0/0 rx:0/0 10  (normal call clearing.)
```

```
 IP 0.0.0.0 AcceptedMime:0 DiscardedMime:0

7    : 6567382hs.35 +1054 +3550 pid:310 Originate 576341.
 tx:11/14411 rx:0/0 10  (Normal connection)
 Telephony : tx:35500/10540/24960ms g729r8 noise:115dBm acom:-14dBm

7    : 6567308hs.36 +1128 +4517 pid:0 Answer
tx:0/0 rx:0/0 10  (normal call clearing.)
 IP 0.0.0.0 AcceptedMime:0 DiscardedMime:0
```

The following is output from the **show call history** command.

```
router# show call history fax 1 2

GENERIC:
SetupTime=23193 ms
Index=1
PeerAddress=527....
PeerSubAddress=
PeerId=3469
PeerIfIndex=157
LogicalIfIndex=30
DisconnectCause=10
DisconnectText=normal call clearing.: Normal connection
ConnectTime=24284
DisconectTime=31288
CallOrigin=1
ChargedUnits=0
InfoType=fax
TransmitPackets=62
TransmitBytes=88047
ReceivePackets=0
ReceiveBytes=0

TELE:
ConnectionId=[0x37EC7F41 0xB0110001 0x0 0x35C34]
TxDuration=80950 ms
FaxTxDuration=10910 ms
FaxRate=14400
NoiseLevel=-1
ACOMLevel=-1
SessionTarget=
ImgPages=3

GENERIC:
SetupTime=22021 ms
Index=2
PeerAddress=wook song
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=24284
DisconectTime=31545
CallOrigin=2
ChargedUnits=0
InfoType=fax
TransmitPackets=0
TransmitBytes=0
ReceivePackets=0
ReceiveBytes=41190
```

```
MMOIP:
ConnectionId[0x37EC7F41 0xB0110001 0x0 0x35C34]
RemoteIPAddress=0.0.0.0
SessionProtocol=SMTP
SessionTarget=
MessageId=
AccountId=
ImgEncodingType=MH
ImgResolution=fine
AcceptedMimeTypes=2
DiscardedMimeTypes=1
Notification=None
```

Table 27 provides an alphabetical listing of the fields for the **show call history** command and a description of each field.

Table 27 show call history field Descriptions

Field	Description
ACOMLevel	Average ACOM level for this call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for a particular call.
CallOrigin	Call origin: answer or originate.
CoderTypeRate	Negotiated coder rate. This value specifies the transmit rate of voice/fax compression to its associated call leg for this call.
ConnectionID	Global call identifier for the gateway call.
ConnectTime	Time this call was connected.
DisconnectCause	Description explaining why this call was disconnected.
DisconnectText	Descriptive text explaining the disconnect reason.
DisconnectTime	Time this call was disconnected.
FaxDuration	Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithRedundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithSilence	Duration of voice signal replaced with silence because the voice data was lost or not received in time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during the voice call.
Index	Dial peer identification number.
InfoType	Information type for this call.

Table 27 show call history field Descriptions (continued)

Field	Description
LogicalIfIndex	Index number of the logical voice port for this call.
LoWaterPlayoutDelay	Low water mark Voice Playout FIFO Delay during the voice call.
NoiseLevel	Average noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
PeerAddress	Destination pattern or number associated with this peer.
PeerId	ID value of the peer entry table to which this call was made.
PeerIfIndex	Index number of the logical interface through which this call was made. For ISDN media, this would be the index number of the B channel used for this call.
PeerSubAddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the Decoder Delay during this voice call.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for this call.
RemoteUDPPort	Remote system UDP listener port to which voice packets are sent.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone for this call.
SelectedQoS	Selected RSVP QoS for this call.
Session Protocol	Session protocol to be used for an Internet call between the local and remote router via the IP backbone.
Session Target	Session target of the peer used for third call.
SetUpTime	Value of the system UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes sent by this peer during this call.
TransmitPackets	Number of packets sent by this peer during this call.
TxDuration	Duration of the transmit path open from this peer to the voice gateway for this call.
VADEnable	Whether voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice sent from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration by the TxDuration value.

Related Commands

Command	Description
show call active	Displays active call information for a fax transmission in progress.

show call history video record

To display information about video calls, use the **show call history video record** command in privileged EXEC mode.

show call history video record

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC.

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for the Cisco MC3810.

Usage Guidelines

Use this command to review statistics about recent incoming and outgoing video calls.

Examples

On a Cisco MC3810, the following example displays information about two video calls:

```
Router# show call history video record

CallId = 4
CalledNumber = 221
CallDuration = 39006 seconds
DisconnectText = remote hangup
SVC: call ID = 8598630
Remote NSAP = 47.0091810000000002F26D4901.00107B09C645.C8
Local NSAP = 47.0091810000000002F26D4901.00107B4832E1.C8
vcd = 414, vpi = 0, vci = 158
SerialPort = Serial0
VideoSlot = 1, VideoPort = 0


CallId = 3
CalledNumber = 221
CallDuration = 557 seconds
DisconnectText = local hangup
SVC: call ID = 8598581
Remote NSAP = 47.0091810000000002F26D4901.00107B09C645.C8
Local NSAP = 47.0091810000000002F26D4901.00107B4832E1.C8
vcd = 364, vpi = 0, vci = 108
SerialPort = Serial0
VideoSlot = 1, VideoPort = 0
```

show call history voice record

To display Call Detail Record (CDR) events in the call history table, use the **show call history voice records** privileged EXEC command.

show call history voice record

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC.

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for the Cisco MC3810.

Examples

The following example displays a sample of voice call history records showing a local call between two telephones attached to the same Cisco MC3810:

```
router# show call history voice record

ConnectionId=[0x2C7AEFDC 0x59830001 0x0 0xB0AAA3]
Media=TELE, TxDuration= 1418 ms
CallingNumber=2001
SetupTime=1157801 x 10ms
ConnectTime=1158046 x 10ms
DisconectTime=1158188 x 10ms
DisconnectText=local onhook

ConnectionId=[0x2C7AEFDC 0x59830001 0x0 0xB0AAA3]
Media=TELE, TxDuration= 1422 ms
CalledNumber=2002
SetupTime=1157802 x 10ms
ConnectTime=1158046 x 10ms
DisconectTime=1158188 x 10ms
DisconnectText=remote onhook
```

Table 28 explains the fields in the sample output.

Table 28 ***show call history voice record Field Descriptions***

Field	Description
ConnectionID	Global call identifier for this voice call
Media	Call over the type of media. If the call is over the (telephone) access side, the entry will be TELE. If the call is over the voice network side, the entry will be either ATM, FR (for Frame Relay), or HDLC.
LowerIFName	Physical lower interface information. Only displays if the media is either ATM, FR, or HDLC.
TxDuration	The length of the call. Only displays if the media is TELE.
CalledNumber	The called number.
CallingNumber	The calling number.
SetupTime	Time the call setup started.
ConnectTime	Time the call is connected.
DisconnectTime	Time the call is disconnected.
DisconnectText	Descriptive text explaining the reason for disconnect.

Related Commands

Command	Description
show call active voice	Displays the Voice over IP active call table.
show dial-peer voice	Displays configuration information for dial peers.
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 call resource voice stats

To displays resource statistics for an H.323 gateway, use the **show call resource voice stats** command in privileged EXEC mode.

show call resource voice stats

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)T	This command was introduced.

Usage Guidelines

This command displays the H.323 resources that are monitored when the **resource threshold** command is used to configure and enable resource threshold reporting.

Examples

The following example shows the resource statistics for an H.323 gateway:

```
gateway1# show call resource voice stats

Resource Monitor -  Dial-up Resource Statistics Information:

DSP Statistics:

Utilization: 0 percent
Total channels: 48
Inuse channels: 0
Disabled channels 0:
Pending channels: 0
Free channels: 48

DS0 Statistics:

Total channels: 0
Addressable channels: 0
Inuse channels: 0
Disabled channels: 0
Free channels: 0
```

Table 29 explains the fields in the sample output.

Table 29 show call resource voice stats Command Field Descriptions

Statistic	Definition
Total channels	Number of physically configured channels for the resource.
Addressable channels	Number of channels that can be used for a specific type of dial-up service, such as H.323 which includes all the DS0 resources that have been associated to a voice POTS dial plan profile.
Inuse channels	Number of addressable channels that are in use. This includes all channels that either have active calls or have been reserved for testing.
Free channels	Number of addressable channels that are free.
Pending channels	Number of addressable channels that are pending in loadware download.
Disabled channels	Number of addressable channels that are physically down or that have been disabled administratively with the shut down or busy out command.

Related Commands

Command	Description
resource threshold	Configures a gateway to report H.323 resource availability to the gatekeeper of the gateway.
show call resource voice threshold	Displays the threshold configuration settings and status for an H.323 gateway.

show call resource voice threshold

To display the threshold configuration settings and status for an H.323 gateway, use the **show call resource voice threshold** command in privileged EXEC mode.

show call resource voice threshold

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)T	This command was introduced.

Usage Guidelines

This command displays the H.323 resource thresholds that are configured with the **resource threshold** command.

Examples

The following example shows the resource threshold settings and status for an H.323 gateway:

```
gateway1# show call resource voice threshold

Resource Monitor -  Dial-up Resource Threshold Information:

DS0 Threshold:

Client Type: h323
High Water Mark: 70
Low Water Mark: 60
Threshold State: init
DSP Threshold:

Client Type: h323
High Water Mark: 70
Low Water Mark: 60
Threshold State: low_threshold_hit
```

Related Commands

Command	Description
resource threshold	Configures a gateway to report H.323 resource availability to the gatekeeper of the gateway
show call resource voice stats	Displays resource statistics for an H.323 gateway.

show cdapi

To display the Call Distributor Application Programming Interface (CDAPI), use the **show cdapi** command in privileged EXEC mode.

show cdapi

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

CDAPI is the internal API that provides an interface between signalling stacks and applications.

Examples

The following is output for the **show cdapi** command:

```
Router# show cdapi

Registered CDAPI Applications/Stacks
====================================
Application TSP CDAPI Application
        Application Type(s)  Voice Facility Signaling
        Application Level    Tunnel
        Application Mode     Enbloc
Signaling Stack ISDN
        Interface Se023
Signaling Stack ISDN
        Interface Se123
Active CDAPI Calls
==================
Interface Se023
        No active calls.
Interface Se123
        Call ID = 0x39, Call Type = VOICE, Application = TSP CDAPI Application
CDAPI Message Buffers
=====================
Used Msg Buffers 0, Free Msg Buffers 1600
Used Raw Buffers 1, Free Raw Buffers 799
Used Large-Raw Buffers 0, Free Large-Raw Buffers 80
scarlatti1#
```

Related Commands

Command	Description
isdn protocol-emulate	Configures the Layer 2 and Layer 3 port protocol of a BRI voice port or a PRI interface to emulate NT (network) or TE (user) functionality.
isdn switch type	Configures the Cisco AS5300 PRI interface to support Q.SIG signalling.
pri-group nec-fusion	Configures your NEC PBX to support FCCS.
show rawmsg	Displays the raw messages owned by the required component.

show connect

To display configuration information about drop-and-insert connections that have been configured on a router, enter the **show connect** command in privileged EXEC mode.

show connect {**all** | **elements** | **name** | **id** | **port** {**T1** | **E1**} *slot/port*}

Syntax Description

all	Shows a table of all configured connections.
elements	Shows registered hardware or software interworking elements.
name	Displays a connection that has been named by using the **connect** global configuration command. The name you enter is case-sensitive and must match the configured name exactly.
id	Displays the status of a connection that you specify by an identification number or range of identification numbers. The router assigns these IDs automatically in the order that they were created, beginning with 1. The **show connect all** command displays these IDs.
port	Displays the status of a connection that you specify by indicating the type of controller (T1 or E1) and location of the interface.
T1	Specifies a T1 controller.
E1	Specifies an E1 controller.
slot/port	The location of the T1 or E1 controller port whose connection status you want to see. Valid values for *slot* and *port* are 0 and 1.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced.

Usage Guidelines

This command shows Drop and Insert connections on the Cisco 2600 and 3600 series.

The command displays different information in different formats depending on the keyword that you use.

Examples

The following examples show how the same tabular information appears when you enter different keywords:

```
Router# show connect all
ID    Name               Segment 1            Segment 2           State
=========================================================================
1     Test             -T1 1/0 01           -T1 1/1 02          ADMIN UP
2     Test2            -T1 1/0 03           -T1 1/1 04          ADMIN UP

Router# show connect id 1-2
ID    Name               Segment 1            Segment 2           State
=========================================================================
1     Test             -T1 1/0 01           -T1 1/1 02          ADMIN UP
2     Test2            -T1 1/0 03           -T1 1/1 04          ADMIN UP

Router# show connect port t1 1/1
ID    Name               Segment 1            Segment 2           State
=========================================================================
1     Test             -T1 1/0 01           -T1 1/1 02          ADMIN UP
2     Test2            -T1 1/0 03           -T1 1/1 04          ADMIN UP
```

The following examples show details about specific connections, including the number of time slots in use and the switching elements:

```
Router# show connect id 2
Connection: 2 - Test2
 Current State: ADMIN UP
 Segment 1: -T1 1/0 03
  TDM timeslots in use: 14-18 (5 total)
 Segment 2: -T1 1/1 04
  TDM timeslots in use: 14-18
Internal Switching Elements: VIC TDM Switch

Router# show connect name Test
 Connection: 1 - Test
 Current State: ADMIN UP
 Segment 1: -T1 1/0 01
  TDM timeslots in use: 1-13 (13 total)
 Segment 2: -T1 1/1 02
  TDM timeslots in use: 1-13
Internal Switching Elements: VIC TDM Switch
```

Related Commands

Command	Description
connect	Defines connections between T1 or E1 controller ports for Drop and Insert.
tdm-group	Configures a list of time slots for creating clear channel groups (pass-through) for TDM cross-connect.

show controllers cable

To display information about the interface controllers for a specific cable modem card slot in a Cisco uBR7200 series cable router, use the **show controllers cable** command in privileged EXEC mode.

show controllers cable *slot/port* [**downstream** | **upstream** [*port*]]

Syntax Description

slot/port	Slot number/port number indicating the location of the Cisco cable modem card.
downstream	(Optional) Displays downstream interface status.
upstream	(Optional) Displays upstream interface status.
port	(Optional) Selects specific upstream port.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 XA	This command was introduced.
12.0(2)XC	This command was modified.

Examples

The following is sample output from the **show controllers cable upstream** command for the cable modem located in slot 4, port 0:

```
CMTS01# show controllers cable 4/0 upstream 2
 Cable4/0 Upstream 2 is administratively down
  Frequency 5.008 MHz, Channel Width 0.200 MHz, QPSK Symbol Rate 0.160 Msps
  Spectrum Group 4
  Nominal Input Power Level 5 dBmV, Tx Timing Offset 0
  Ranging Backoff Start 16, Ranging Backoff End 16, Tx Backoff Start 16
  Tx Backoff End 16, Modulation Profile Group 1
  part_id=0x3137, rev_id=0x01, rev2_id=0xFF
  nb_agc_thr=0x0000, nb_agc_nom=0x0000
  Range Load Reg Size=0x58
  Request Load Reg Size=0x0E
  Minislot Size in number of Timebase Ticks is = 8
  Minislot Size in Symbols =8
  Bandwidth Requests = 0x0
  Piggyback Requests = 0x0
  Invalid BW Requests= 0x0
  Minislots Requested= 0x0
  Minislots Granted  = 0x0
  Minislot Size in Bytes = 2
  UCD Count = 0
  DES Ctrl Reg#0 = C00C0C43, Reg#1 = 0
```

Table 30 describes the fields shown in the **show controllers cable upstream** display.

Table 30 show controllers cable upstream Command Field Descriptions

Field	Description
Cable	Slot number/port number indicating the location of the Cisco cable modem card.
Upstream is administratively down	Indicates the RF upstream interface is disabled.
Frequency	Transmission frequency of the RF upstream channel.
Channel Width	Indicates the width of the RF upstream channel.
QPSK Symbol Rate	Indicates the modulation technique for upstream transmission.
Spectrum Group 4	Indicates the spectrum group associated with this slot and port.
Nominal Input Power level	Indicates the desired power level coming into the receiver.
Tx Timing Offset	Indicates the current ranging offset on the channel.
Ranging Backoff Start	Indicates how many ranging slots to back off before resending the ranging bursts after an upstream collision. Expressed as exponents of 2. See Ranging Backoff End.
Ranging Backoff End	Indicates how many ranging slots to back off before resending the ranging bursts after an upstream collision. Expressed as exponents of 2. See Ranging Backoff Start.
Tx Backoff Start	Indicates the starting exponential backoff value for data collisions.
Tx Backoff End	Indicates the ending exponential backoff value for data collisions.
Modulation Profile Group	A set of burst profiles defining an upstream range.
part_id=	The part number of the Phy chip. FFFF means the Phy chip is turned off.
rev_id=	The Phy chip revision number.
rev2_id=	The Phy chip sub-revision number.
nb_agc_thr=	Threshold used to control gain.
nb_agc_nom=	Used to accelerate convergence of input power level.
Range Load Reg Size=	Size, indicated by number of symbols, for range request bursts.
Request Load Reg Size=	Size, indicated by number of symbols, for request bursts.
Minislot Size in number of Timebase Ticks is	Size in tick units of upstream minislot. A tick is 6.25 microseconds.
Minislot Size in Symbols	Size in symbols of the upstream minislot.
Bandwidth Requests	Number of successful bandwidth requests received in the contention minislots.
Piggyback Requests	Number of successful bandwidth requests piggybacked with regular data transmissions.
Invalid BW Requests	Number of invalid bandwidth (BW) requests. (An example of an invalid bandwidth request is a modem using a non-existent SID to request bandwidth.
Minislots Requested	Total number of minislots requested.

Table 30 show controllers cable upstream Command Field Descriptions (continued)

Field	Description
Minislots Granted	Total number of minislots granted.
Minislot Size in Bytes	Size of the minislot in bytes.
UCD Count	Number of UCDs sent for this upstream.
DES Ctrl Reg # =	Interval DES controller register dump.

The following is sample output for the downstream connection for slot 3 on port 0 from the **show controllers cable downstream** command:

```
CMTS01# show controllers cable 3/0 downstream
Cable 3/0 Downstream is up
Frequency not set, Channel Width 6 MHz, 64-QAM,
Symbol Rate 5.056941 Msps
FEC ITU-T J.83 Annex A, R/S Interleave I=12, J=17
```

Table 31 describes the fields shown in the **show controllers cable downstream** display.

Table 31 show controllers cable downstream Field Descriptions

Field	Description
Cable	Slot number/port number indicating the location of the Cisco cable modem card.
Downstream is up	Indicates the RF downstream interface is enabled.
Frequency	Transmission frequency of the RF downstream. (This information may not match the current transmission frequency, which is external to uBR.)
Channel Width	Indicates the width of the RF downstream channel.
QAM	Indicates the modulation scheme.
Symbol Rate	Indicates the transmission rate (in number of symbols per second).
FEC ITU-T	Indicates the MPEG framing standard.
R/S Interleave I/J	Indicates Reed Solomon framing based on ITU S.83-B.

Related Commands

Command	Description
show interface cable sid	Displays information about interface controllers for a specific cable access router card slot.

show controllers rs366

To display information about the RS-366 video interface on the video dialing module (VDM), use the **show controllers rs366** command in privileged EXEC mode.

show controllers rs366 *slot port*

Syntax Description

slot	Slot location of the VDM module. On the Cisco MC3810, this value is either 1 or 2. If you do not enter the correct location, the command is rejected.
port	Port location of the RS-366 interface in the VDM module. On the Cisco MC3810, this value is 0.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for the Cisco MC3810.

Examples

On a Cisco MC3810, the following example displays information about the RS-366 controller:

```
Router# show controller rs366 0 1

RS366:driver is initialized in slot 1, port 0:

STATUS STATE LSR  LCR  ICSR EXT  T1     T2     T3     T4     T5
0x02   0x01  0x00 0x50 0xE0 0x00 5000   5000   5000   20000  10000
Dial string:
121C
```

Table 32 explains the meaning of the fields in the **show controllers rs366** command.

Table 32 show controllers cable downstream Field Descriptions

Field	Description
STATUS	Last interrupt status.
STATE	Current state of the state machine.
LSR	Line status register of the VDM.
LCR	Line control register of the VDM.
ICSR	Interrupt control and status register of the VDM.
EXT	Extended register of the VDM.

Table 32 show controllers cable downstream Field Descriptions (continued)

Field	Description
T1 through T5	Timeouts 1 through 5 of the watchdog timer in milliseconds.
Dial string	Most recently dialed number collected by the driver. 0xC at the end of the string indicates the EON (end of number) character.

show controllers voice

To display information about voice-related hardware, use the **show controllers voice** command in Privileged EXEC mode.

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)XQ	This command was introduced.

Usage Guidelines

This command displays interface status information that is specific to voice-related hardware, such as, the registers of the TDM switch, the host port interface of the digital signal processor (DSP), and the DSP firmware versions. The information displayed is generally only useful for diagnostic tasks performed by technical support.

Examples

The following is an example of the output from the **show controllers voice** command:

```
router# show controllers voice
EPIC Switch registers:
STDA 0xFF STDB 0xFF SARA 0xAD SARB 0xFF SAXA 0xFF SAXB 0x0 STCR 0x3F
MFAIR 0x3F
STAR 0x65 OMDR 0xE2 VNSR 0x0 PMOD 0x4C PBNR 0xFF POFD 0xF0 POFU 0x18
PCSR 0x1 PICM 0x0 CMD1 0xA0 CMD2 0x70 CBNR 0xFF CTAR 0x2 CBSR 0x20 CSCR
0x0

DSP 0 Host Port Interface:
HPI Control Register 0x202
InterfaceStatus 0x2A MaxMessageSize 0x80
RxRingBufferSize 0x6 TxRingBufferSize 0x9
pInsertRx 0x4 pRemoveRx 0x4 pInsertTx 0x6 pRemoveTx 0x6

Rx Message 0:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:   0000 4AC7 5F08 91D1 0000 0000 7DF1 69E5 63E1 63E2
0020:   6E7C ED67 DE5D DB5C DC60 EC7E 6BE1 58D3 50CD 4DCE
0040:   50D2 5AE5 7868 DA52 CE4A C746 C647 C94B D25A EAF4
0060:   5DD7 4FCD 4ACA 4ACC 4FD3 5DE8 F769 DC58 D352 D253
0080:   D65B E573 6CDF 59D3 4ECF 4FD0

Rx Message 1:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:   0000 1CDD 3E48 3B74 0000 0000 3437 3D4C F0C8 BBB5
0020:   B2B3 B7BF D25B 4138 3331 3339 435F CFBD B6B2 B1B4
0040:   BBC8 7E48 3B34 3131 363D 4FDE C3B9 B3B1 B3B8 C2DB
```

```
0060:    533F 3833 3235 3B48 71CC BDB7 B4B5 B8BF CF67 483D
0080:    3836 383C 455B DAC6 BDB9 B9BB

Rx Message 2:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:    0000 4AC8 5F08 9221 0000 0000 54DA 61F5 EF60 DA53
0020:    CF4F CD4E D256 DB63 FCEE 5FDA 55D1 50CF 4FD3 56D8
0040:    5DE1 6E7C EC60 DC59 D655 D456 D85D DF6A F4F4 69E2
0060:    5CDD 5BDC 5BDE 61E9 6DF1 FF76 F16D E96A E566 EA6A
0080:    EB6F F16D EF79 F776 F5F5 73F0

Rx Message 3:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:    0000 1CDE 3E48 3BC4 0000 0000 C0CC EC54 453E 3C3C
0020:    3F47 56F3 D1C7 C1BF C0C6 CEE1 6752 4A46 4648 4E59
0040:    6FE4 D6CF CDCE D2DA E57E 675E 5B5B 5E62 6B76 FCF6
0060:    F6FA 7D75 7373 7BF5 EAE1 DCDA DADD E6FE 6559 514D
0080:    4D4E 5563 EFD9 CDC8 C5C6 CAD1

Rx Message 4:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:    0000 4AC6 5F08 9181 0000 0000 DD5B DC5E E161 E468
0020:    FAFD 6CE1 5AD3 53D1 53D7 61EC EA59 CF4A C644 C344
0040:    CA4E D86C 60D0 48C2 3EBD 3CBD 3EC0 47CF 5976 DF4F
0060:    C945 C242 C146 C94E D668 73DB 54CE 4DCC 4DCE 53DB
0080:    64F9 ED63 DC59 DA58 DC5D E46C

Rx Message 5:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:    0000 1CDC 3E48 3B24 0000 0000 5B5B 5D62 6A76 FCF5
0020:    F5F9 7D78 7374 7CF5 EAE1 DDDA DBDD E7FE 6559 514E
0040:    4D4F 5663 EFD8 CDC8 C6C6 CAD1 E760 4E46 403F 4047
0060:    5173 D5C7 BFBC BCBE C5D4 6D4C 3F3B 3939 3D46 5ADB
0080:    C5BC B7B6 B8BD C8E8 4F3F 3835

Tx Message 0:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:    0000 4AC6 5F08 9181 0000 003C DD5B DC5E E161 E468
0020:    FAFD 6CE1 5AD3 53D1 53D7 61EC EA59 CF4A C644 C344
0040:    CA4E D86C 60D0 48C2 3EBD 3CBD 3EC0 47CF 5976 DF4F
0060:    C945 C242 C146 C94E D668 73DB 54CE 4DCC 4DCE 53DB
0080:    64F9 ED63 DC59 DA58 DC5D E46C

Tx Message 1:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:    0000 1CDC 3E48 3B24 0000 003C 5B5B 5D62 6A76 FCF5
0020:    F5F9 7D78 7374 7CF5 EAE1 DDDA DBDD E7FE 6559 514E
0040:    4D4F 5663 EFD8 CDC8 C6C6 CAD1 E760 4E46 403F 4047
0060:    5173 D5C7 BFBC BCBE C5D4 6D4C 3F3B 3939 3D46 5ADB
0080:    C5BC B7B6 B8BD C8E8 4F3F 3835

Tx Message 2:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:    0000 4AC7 5F08 91D1 0000 003C 7DF1 69E5 63E1 63E2
0020:    6E7C ED67 DE5D DB5C DC60 EC7E 6BE1 58D3 50CD 4DCE
0040:    50D2 5AE5 7868 DA52 CE4A C746 C647 C94B D25A EAF4
0060:    5DD7 4FCD 4ACA 4ACC 4FD3 5DE8 F769 DC58 D352 D253
0080:    D65B E573 6CDF 59D3 4ECF 4FD0

Tx Message 3:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:    0000 1CDD 3E48 3B74 0000 003C 3437 3D4C F0C8 BBB5
0020:    B2B3 B7BF D25B 4138 3331 3339 435F CFBD B6B2 B1B4
0040:    BBC8 7E48 3B34 3131 363D 4FDE C3B9 B3B1 B3B8 C2DB
```

```
0060:    533F 3833 3235 3B48 71CC BDB7 B4B5 B8BF CF67 483D
0080:    3836 383C 455B DAC6 BDB9 B9BB

Tx Message 4:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:    0000 4AC8 5F08 9221 0000 003C 54DA 61F5 EF60 DA53
0020:    CF4F CD4E D256 DB63 FCEE 5FDA 55D1 50CF 4FD3 56D8
0040:    5DE1 6E7C EC60 DC59 D655 D456 D85D DF6A F4F4 69E2
0060:    5CDD 5BDC 5BDE 61E9 6DF1 FF76 F16D E96A E566 EA6A
0080:    EB6F F16D EF79 F776 F5F5 73F0

Tx Message 5:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:    0000 1CDE 3E48 3BC4 0000 003C C0CC EC54 453E 3C3C
0020:    3F47 56F3 D1C7 C1BF C0C6 CEE1 6752 4A46 4648 4E59
0040:    6FE4 D6CF CDCE D2DA E57E 675E 5B5B 5E62 6B76 FCF6
0060:    F6FA 7D75 7373 7BF5 EAE1 DCDA DADD E6FE 6559 514D
0080:    4D4E 5563 EFD9 CDC8 C5C6 CAD1

Tx Message 6:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:    0000 1CDA 3E48 3A84 0000 003C E75F 4E46 403F 4147
0020:    5174 D5C7 BFBC BCBE C5D4 6C4C 3F3B 3939 3D46 5BDA
0040:    C5BC B7B6 B8BD C8E9 4F3F 3834 3437 3D4C EEC8 BBB5
0060:    B2B3 B8BF D35A 4138 3331 3339 435F CEBD B6B1 B1B4
0080:    BBC9 7C48 3B34 3131 363D 4FDE

Tx Message 7:
packet_length 100 channel_id 1 packet_id 0 process id 0x1
0000:    0000 4AC5 5F08 9131 0000 003C 66DE 66EB 67EE FE6E
0020:    F7E7 6B68 E068 EE6A DF5C DF62 EDF1 6FF2 7A78 67DC
0040:    5EDF 62E7 64E6 66E0 7071 EA69 F86E E260 DE5D E665
0060:    EB75 F0FB 6DE9 64E4 69E3 66EA 67E9 6DF9 F177 EC6E
0080:    EB6E F876 F875 7D6E E966 E05D

Tx Message 8:
packet_length 100 channel_id 2 packet_id 0 process id 0x1
0000:    0000 1CDB 3E48 3AD4 0000 003C C2B9 B3B1 B3B8 C2DC
0020:    523F 3733 3235 3C49 72CB BDB7 B4B5 B8BF CF67 483C
0040:    3836 373C 455C DAC6 BDB9 B9BB C0CC EE54 453E 3C3C
0060:    3F47 56F1 D1C7 C1BF C0C6 CEE1 6651 4A46 4648 4D59
0080:    70E3 D6CF CDCE D2D9 E67E 675E

Bootloader 1.8, Appn 3.1
Application firmware 3.1.8, Built by claux on Thu Jun 17 11:00:05 1999

VIC Interface Foreign Exchange Station 0/0, DSP instance (0x19543C0)
Singalling channel num 128 Signalling proxy 0x0 Signaling dsp 0x19543C0
tx outstanding 0, max tx outstanding 32
ptr 0x0, length 0x0, max length 0x0
dsp_number 0, Channel ID 1
received 0 packets, 0 bytes, 0 gaint packets
0 drops, 0 no buffers, 0 input errors 0 input overruns
650070 bytes output, 4976 frames output, 0 output errors, 0 output
underrun
0 unaligned frames

VIC Interface Foreign Exchange Station 0/1, DSP instance (0x1954604)
Singalling channel num 129 Signalling proxy 0x0 Signaling dsp 0x1954604
tx outstanding 0, max tx outstanding 32
ptr 0x0, length 0x0, max length 0x0
dsp_number 0, Channel ID 2
received 0 packets, 0 bytes, 0 gaint packets
0 drops, 0 no buffers, 0 input errors 0 input overruns
```

```
393976 bytes output, 3982 frames output, 0 output errors, 0 output
underrun
0 unaligned frames
```

Related Commands

Command	Description
show dial-peer voice	Displays configuration information and call statistics for dial peers.
show interface dspfarm dsp	Displays hardware information including DRAM, SRAM, and the revision-level information on the line card.
show voice dsp	Displays the current status of all DSP voice channels on the Cisco MC3810 multiservice concentrator.
show voice port	Displays configuration information about a specific voice port.

show csm

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

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 33 explains the fields contained in both of these examples.

Table 33 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 33 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 33 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.

Table 33 show csm voice Field Descriptions (continued)

Field	Description
wdt_timeout_count=	Number of times the watchdog timer is activated for this call.
wdt_timestamp_starte d	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.
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_time slot =	Total number of free RBS (CAS) time slots available for the whole system.
total_busy_rbs_time slot =	Total number of RBS (CAS) time slots that have been busied out. This includes both dynamically and statically busied out RBS time slots.

Table 33 show csm voice Field Descriptions (continued)

Field	Description
total_dynamic_busy_rbs_time slot =	Total number of RBS (CAS) time slots that have been dynamically busied out.
total_static_busy_rbs_time slot =	Total number of RBS (CAS) time slots 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 contents of the active call table.
show call history voice	Displays the contents of the call history table.
show num-exp	Displays how number expansions are configured.
show voice port	Displays configuration information about a specific voice port.

show dhcp

To display the current Dynamic Host Configuration Protocol (DHCP) settings on point-to-point interfaces, use the **show dhcp** command in privileged EXEC mode.

show dhcp {**lease** | **server**}

Syntax Description

lease	Displays DHCP addresses leased from a server.
server	Displays known DHCP servers.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

You can use this command on any point-to-point type of interface that uses DHCP for temporary IP address allocation.

Examples

The following is sample output for the **show dhcp lease** command:

```
uBR924# show dhcp lease
Temp IP addr: 188.188.1.40  for peer on Interface: cable-modem0
Temp sub net mask: 0.0.0.0
   DHCP Lease server: 4.0.0.32, state: 3 Bound
   DHCP transaction id: 2431
   Lease: 3600 secs,  Renewal: 1800 secs,  Rebind: 3150 secs
Temp default-gateway addr: 188.188.1.1
   Next timer fires after: 00:58:01
   Retry count: 0   Client-ID: 0010.7b43.aa01
```

Table 34 describes the significant fields shown in the display.

Table 34 show dhcp lease Field Descriptions

Field	Description
Temp IP addr	IP address leased from the DHCP server for the cable access router interface.
Temp subnet mask	Temporary subnet mask assigned to the cable access router interface.
DHCP Lease server	IP address of the DHCP server that assigned an IP address to this client.

Table 34 show dhcp lease Field Descriptions (continued)

Field	Description
state	Current state of this client (the cable access router interface). Possible states are Bound, Renew, or Rebinding. For descriptions of these states, see RFC 2131.
DHCP transaction id	Unique number established by the Cisco uBR924 before the first request message is sent to the DHCP server. The same transaction id is used as long as the lease keeps getting renewed and is valid. If a new "discover" message is sent, a new transaction ID is used.
Lease	Time (in seconds) for which the leased IP address is valid; the duration of the lease.
Renewal	Time interval (in seconds) from address assignment until the client transitions to the renewing state. When the renewal (T1) time expires, the client sends a unicast dhcprequest message to the server to extends its lease. The default value of this timer is 0.5 times the duration of the lease.
Rebind	Time interval (in seconds) from address assignment until the client transitions to the rebinding state and sends a broadcast dhcprequest message to any DHCP server to extends its lease. The default value of this timer (T2) is 0.875 times the duration of the lease.
Temp default-gateway addr	IP address of the router closest to this client on the network.
Next timer fires after	Time in hours, minutes, and seconds until the next timer expires.
Retry count	Number of times the client has sent any message to the DHCP server—most likely a request message to extend its lease. When the lease is renewed, the Retry count is reset to 0.
Client-ID	MAC address (with optional media type code) that uniquely identifies the client on the subnet for binding lookups.

The following is sample output for the **show dhcp server** command:

```
uBR924# show dhcp server
   DHCP server: ANY (255.255.255.255)
    Leases:    1
    Offers:    1      Requests: 2      Acks: 1      Naks: 0
    Declines: 0       Releases: 0      Bad:  0
    TFTP Server Name: SOHOSERVER
    TIME0: 1.2.0.250,  TIME1: 0.0.0.0
    Subnet: 255.255.255.0
```

Table 35 describes the significant fields shown in the display.

Table 35 show dhcp server Field Descriptions

Field	Description
DHCP server	MAC address used by the DHCP server.
Leases	Number of current leased IP addresses.
Offers	Number of offers for an IP address sent to a proxy client from the server.
Requests	Number of requests for an IP address to the server.

Table 35 show dhcp server Field Descriptions (continued)

Field	Description
Acks	Number of acknowledge messages sent by the server to the proxy client.
Naks	Number of not acknowledge ' messages sent by the server to the proxy client.
Declines	Number of offers from the server that have been declined by the proxy client.
Releases	Number of times IP addresses have been relinquished gracefully by the client.
Bad	Number of bad packets received due to wrong length, wrong field type, or other causes.
TFTP Server Name	Name (if any) configured for the server providing TFTP downloads to the cable modem.
TIME0	IP address of the primary Time of Day (ToD) server.
TIME1	IP address of the secondary Time of Day (ToD) server.
Subnet	Subnet containing the DHCP server.

Related Commands

Command	Description
cable-modem voip best-effort	Allows voice calls to be sent upstream over the cable interface via best effort.
show bridge cable-modem	Displays bridging information for a cable modem.
show interfaces cable-modem	Displays information about the cable interface of the Cisco uBR900 series cable access router.

show diag

To display the revision level information for a Cisco uBR7200 series cable modem card, use the **show diag** command in privileged EXEC mode.

show diag

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.1 CA	This command was introduced.
11.2 P	This command was modified to update the sample display for the port adapters PA-12E/2FE, PA-E3, and PA-T3.
11.3 XA	This command was made available for Cisco IOS Release 11.3 XA.
12.0(5)XQ	This command was enhanced and made available for the Cisco 1750 router.

Usage Guidelines

This command displays information for the electrically erasable programmable read-only memory (EEPROM), the motherboard, and the WAN interface cards and voice interface cards (WICs and VICs).

Examples

The following is sample output from the **show diag** command displaying revision level information for the cable line card (slot 6):

```
CMTS01# show diag
Slot 6:
        MC11 port adapter, 1 port
        Port adapter is analyzed
        Port adapter insertion time 02:37:10 ago
        Hardware Revision        : 1.2
        Part Number              : 800-02455-02
        Board Revision           : 03
        Deviation Number         : 0-3
        Fab Version              : 03
        PCB Serial Number        : 00004500239
        RMA Test History         : 00
        RMA Number               : 0-0-0-0
        RMA History              : 00
        Calibration Data         : Minimum: -8 dBmV, Maximum: 8 dBmV
              Calibration values : 0x5D43 0x3F05 0x1794
        Unknown Field (type 0083): 83 FF FF FF
        EEPROM format version 4
        EEPROM contents (hex):
          0x00: 04 FF 40 00 F1 41 01 02 C0 46 03 20 00 09 97 02
```

```
0x10: 42 30 33 80 00 00 00 03 02 03 C1 8B 30 30 30 30
0x20: 34 35 30 30 32 33 39 03 00 81 00 00 00 00 04 00
0x30: C8 09 F8 08 03 5D 43 3F 05 17 94 83 83 FF FF FF
0x40: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
0x50: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
0x60: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
0x70: FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
```

Table 36 describes the fields shown in the **show diag** display.

Table 36 show diag Field Descriptions

Field	Description
MC11 port adapter	Line card type.
Port adapter is analyzed	The system has identified the Cisco uBR7200 series port adapter.
Port adapter insertion time	Elapsed time since insertion.
Hardware Revision	Version number of the Cisco uBR7200 series port adapter.
Part Number	In the Cisco uBR 7200 series, the part number of the port adapter.
Board Revision	Revision number (signifying a minor revision) of the Cisco uBR 7200 series port adapter.
Deviation Number	Revision number (signifying a minor deviation) of the Cisco uBR7200 series port adapter.
Fab Version	Manufacturing fabrication version number.
PCB Serial Number	Serial number of the printed circuit board.
RMA Test History	Counter indicating how many times diagnostics have been performed on this port adapter.
RMA Number	Return material authorization number, which is an administrative number assigned if port adapter needs to be returned for repair.
RMA History	Counter indicating how many times the port adapter has been returned and repaired.
Calibration Data	Input power calibration range.
Calibration values	Upstream port gain calibration constant.
Unknown Field (type)	Unrecognized EEPROM fields.
EEPROM format version	Version number of the EEPROM format.
EEPROM contents (hex)	Dumps of EEPROM programmed data.

Related Commands

Command	Description
show dial-peer voice	Displays configuration information and call statistics for dial peers.
show voice dsp	Displays the current status of all DSP voice channels on the Cisco MC3810 multiservice concentrator.
show voice port	Displays configuration information about a specific voice port.

show dial-peer video

To display dial-peer configuration, use the **show dial-peer video** command in privileged EXEC mode.

show dial-peer video [*number*] [**summary**]

Syntax Description

number	(Optional) A specific video dial peer. This option displays configuration information for a single dial peer identified by the argument *number*. Valid entries are any integers that identify a specific dial peer, from 1 to 32767.
summary	(Optional) Displays a summary of all video dial-peer information.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for the Cisco MC3810.

Usage Guidelines

Use this command to review video dial-peer configuration.

Examples

On a Cisco MC3810, the following example displays detailed information about all configured video dial peers:

```
Router# show dial-peer video
Video Dial-Peer 1
      type = videocodec, destination-pattern = 111
      port signal = 1/0, port media = Serial1
      nsap = 47.0091810000000050E201B101.00107B09C6F2.C8
Video Dial-Peer 2
      type = videoatm,   destination-pattern = 222
      session-target = ATM0 svc nsap 47.0091810000000050E201B101.00E01E92ADC2.C8
Video Dial-Peer 3
      type = videoatm,   destination-pattern = 333
      session-target = ATM0 pvc 70/70
```

show dial-peer voice

To display configuration information for dial peers, use the **show dial-peer voice** command in privileged EXEC mode.

show dial-peer voice [*number*] [**summary**]

Syntax Description

number	(Optional) A specific dial peer. This option displays configuration information for a single dial peer identified by the *number* argument. Valid entries are any integers that identify a specific dial peer, from 1 to 32767.
summary	(Optional for the Cisco MC3810 only) Displays a summary of all voice dial peers.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA	The **summary** keyword was added for the Cisco MC3810.
12.0(3)XG	This command was modified to support VoFR for the Cisco 2600 series and 3600 series routers.
12.0(4)T	Support was added for VoFR for the Cisco 7200 series routers.

Usage Guidelines

Use the **show dial-peer voice** privileged EXEC command to display the configuration for all VoIP and POTS dial peers configured for the router. To show configuration information for only one specific dial peer, use the argument *number* to identify the dial peer.

Examples

The following is sample output from the **show dial-peer voice** command for a POTS dial peer:

```
router# show dial-peer voice 1
VoiceEncapPeer1
        tag = 1, dest-pat = `+14085291000',
        answer-address = `',
        group = 0, Admin state is up, Operation state is down
        Permission is Both,
        type = pots, prefix = `',
        session-target = `', voice port =
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is ""
        Last Disconnect Text is ""
        Last Setup Time = 0
```

The following is sample output from the **show dial-peer voice** command for a VoIP dial peer:

```
router# show dial-peer voice 10
VoiceOverIpPeer10
        tag = 10, dest-pat = `',
        incall-number = `+14087',
        group = 0, Admin state is up, Operation state is down
        Permission is Answer,
        type = voip, session-target = `',
        sess-proto = cisco, req-qos = bestEffort,
        acc-qos = bestEffort,
        fax-rate = voice, codec = g729r8,
        Expect factor = 10,Icpif = 30, VAD = disabled, Poor QOV Trap = disabled,
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is ""
        Last Disconnect Text is ""
        Last Setup Time = 0
```

Table 37 explains the fields contained in both of these examples.

Table 37 show dial-peer voice Field Descriptions

Field	Description
Accepted Calls	Number of calls from this peer accepted since system startup.
acc-qos	Lowest acceptable quality of service configured for calls for this peer.
Admin state	Administrative state of this peer.
Charged Units	Total number of charging units applying to this peer since system startup. The unit of measure for this field is in hundredths of seconds.
codec	Default voice coder rate of speech for this peer.
Connect Time	Accumulated connect time to the peer since system startup for both incoming and outgoing calls. The unit of measure for this field is in hundredths of seconds.
dest-pat	Destination pattern (telephone number) for this peer.
Expect factor	User-requested Expectation Factor of voice quality for calls via this peer.
fax-rate	Fax transmission rate configured for this peer.
Failed Calls	Number of failed call attempts to this peer since system startup.
group	Group number associated with this peer.
ICPIF	Configured Calculated Planning Impairment Factor (ICPIF) value for calls sent by a dial peer.
incall-number	Full E.164 telephone number to be used to identify the dial peer.
Last Disconnect Cause	Encoded network cause associated with the last call. This value will be updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface.
Last Disconnect Text	ASCII text describing the reason for the last call termination.
Last Setup Time	Value of the System Up Time when the last call to this peer was started.
Operation state	Operational state of this peer.
Permission	Configured permission level for this peer.

Table 37 show dial-peer voice Field Descriptions (continued)

Field	Description
Poor QOV Trap	Whether Poor Quality of Voice trap messages have been enabled or disabled.
Refused Calls	Number of calls from this peer refused since system startup.
req-qos	Configured requested quality of service for calls for this dial peer.
session-target	Session target of this peer.
sess-proto	Session protocol to be used for Internet calls between local and remote router via the IP backbone.
Successful Calls	Number of completed calls to this peer.
tag	Unique dial peer ID number.
VAD	Whether or not voice activation detection (VAD) is enabled for this dial peer.

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 dialplan incall number

To pair different voice ports and telephone numbers for troubleshooting, use the **show dialplan incall number** command in privileged EXEC mode.

show dialplan incall *slot-number/subunit-number/port* **number** *dial string*

Syntax Description

slot-number	Slot number in the Cisco router where the voice network module is installed. Valid entries are from 0 to 3, depending on the voice interface card you have installed.
subunit-number	Subunit on the voice network module where the voice port is located. Valid entries are 0 or 1.
port	Voice port. Valid entries are 0 or 1.
dial string	Particular destination pattern (telephone number).

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Occasionally, an incoming call cannot be matched to a dial peer in the dial peer database. One reason this might occur is that the specified destination cannot be reached via the voice interface through which the incoming call came. Use the **show dialplan incall number** command as a troubleshooting method to resolve the call destination by pairing voice ports and telephone numbers together until there is a match.

Examples

The following example tests whether the telephone extension 57681 can be reached through voice port 1/0/1:

```
show dialplan incall 1/0/1 number 57681
```

Related Commands

Command	Description
show dialplan number	Displays which dial peer is reached when a particular telephone number is dialed.

show dialplan number

To show which dial peer is reached when a particular telephone number is dialed, use the **show dialplan number** command in privileged EXEC mode.

show dialplan number *dial string*

Syntax Description

dial string	Particular destination pattern (telephone number).

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

The **show dialplan number** command is used to test if the dial-plan configuration is valid and working as expected.

Examples

The following example displays the dial peer associated with the destination pattern of 54567:

```
router# show dialplan number 51234

Macro Exp.: 14085551234
VoiceOverIpPeer1004
        tag = 1004, destination-pattern = `+1408555....',
        answer-address = `',
        group = 1004, Admin state is up, Operation state is up
        type = voip, session-target = `ipv4:1.13.24.0',
        ip precedence: 0        UDP checksum = disabled
        session-protocol = cisco, req-qos = best-effort,
        acc-qos = best-effort,
        fax-rate = voice, codec = g729r8,
        Expect factor = 10, Icpif = 30,
        VAD = enabled, Poor QOV Trap = disabled
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is ""
        Last Disconnect Text is ""
        Last Setup Time = 0
Matched: +14085551234   Digits: 7
Target: ipv4:172.13.24.0
```

Table 38 explains the fields contained in this example.

Table 38 show dialplan number Field Descriptions

Field	Description
Macro Exp.	Expected destination pattern for this dial peer.
VoiceOverIpPeer	Identifies the dial peer associated with the destination pattern entered.
tag	Unique dial peer identifying number.
destination-pattern	Destination pattern (telephone number) configured for this dial peer.
answer-address	Answer address configured for this dial peer.
Admin state	Describes the administrative state of this dial peer.
Operation state	Describes the operational state of the dial peer.
type	Type of dial peer (POTS or VoIP).
session-target	Displays the configures session target (IP address or host name) for this dial peer.
ip precedence	Displays the numeric value for the IP Precedence configured for this dial peer.
UDP checksum	Indicates the status of the UDP checksum feature.
session-protocol	Session protocol to be used for Internet calls between local and remote router via the IP backbone.
req-qos	Configured requested quality of service for calls for this dial peer.
acc-qos	Configures acceptable quality of service for calls for this dial peer.
fax-rate	Configured facsimile transmission speed for with this dial peer.
codec	Codec type configured for this dial peer.
Expect factor	Configured value at which the system will generate an SMTP message alerting that the voice quality has dropped.
Icpif	Configured Calculated Planning Impairment Factor (ICPIF) value for calls sent by a dial peer.
VAD	Whether or not voice activation detection (VAD) is enabled for this dial peer.
Poor QOV Trap	Whether Poor Quality of Voice trap messages have been enabled or disabled.
Connect Time	Unit of measure indicating the call connection time associated with this dial peer.
Charged Units	Number of call units charged to this dial peer.
Successful Calls	Number of completed calls to this peer since system startup.
Failed Calls	Number of uncompleted (failed) calls to this peer since system startup.
Accepted Calls	Number of calls from this peer accepted since system startup.
Refused Calls	Number of calls from this peer refused since system startup.
Last Disconnect Cause	Encoded network cause associated with the last call. This value will be updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface.

Table 38 show dialplan number Field Descriptions (continued)

Field	Description
Last Disconnect Text	ASCII text describing the reason for the last call termination.
Last Setup Time	Value of the System Up Time when the last call to this peer was started.
Matched	Destination pattern matched for this dial peer.
Target	Matched session target (IP address or host name) for this dial peer.

Related Commands

Command	Description
show dialplan incall number	Pairs different voice ports and telephone numbers together for troubleshooting Voice over IP.

show frame-relay vofr

To display information about the FRF.11 subchannels being used on Voice over Frame Relay (VoFR) data link controller identifiers (DLCIs), use the **show frame-relay vofr** command in privileged EXEC mode.

show frame-relay vofr [*interface* [*dlci* [*cid*]]]

Syntax Description

interface	(Optional) The specific interface type and number for which you wish to display FRF.11 subchannel information.
dlci	(Optional) The specific data link connection identifier for which you wish to display FRF.11 subchannel information.
cid	(Optional) The specific subchannel for which you wish to display information.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

If this command is entered without specifying an interface, FRF.11 subchannel information will be displayed for all VoFR interfaces and DLCIs configured on the router.

Note This command is currently not supported on the Cisco MC3810 for PVCs configured with the **vofr cisco** command or the **frame-relay interface-dlci voice-encap** command.

Examples

The following is sample output from the **show frame-relay vofr** command when an interface is not specified:

```
3640_vofr# show frame-relay vofr
interface        vofr-type    dlci    cid    cid-type
Serial0/0.1      VoFR         16      4      data
Serial0/0.1      VoFR         16      5      call-control
Serial0/0.1      VoFR         16      10     voice
Serial0/1.1      VoFR cisco   17      4      data
```

The following is sample output from the **show frame-relay vofr** command when an interface is specified:

```
3640_vofr# show frame-relay vofr serial0
interface          vofr-type   dlci   cid   cid-type
Serial0            VoFR        16     4     data
Serial0            VoFR        16     5     call-control
Serial0            VoFR        16     10    voice
```

The following is sample output from the **show frame-relay vofr** command when an interface and a DLCI are specified:

```
3640_vofr# show frame-relay vofr serial0 16
VoFR Configuration for interface Serial0

dlci  vofr-type  cid  cid-type            input-pkts     output-pkts    dropped-pkts
16    VoFR       4    data                0              0              0
16    VoFR       5    call-control        85982          86099          0
16    VoFR       10   voice               2172293        6370815        0
```

The following is sample output from the **show frame-relay vofr** command when an interface, a DLCI, and a CID are specified:

```
3640_vofr# show frame-relay vofr serial0 16 10
VoFR Configuration for interface Serial0 dlci 16

  vofr-type  VoFR      cid 10      cid-type voice
  input-pkts 2172293   output-pkts 6370815   dropped-pkts 0
```

Table 39 describes the fields shown in the display.

Table 39 show frame-relay vofr Field Descriptions

Field	Description
interface	Number of the interface that has been selected for observation of FRF.11 subchannels.
vofr-type	Type of the VoFR DLCI being observed.
cid	The portion of the specified DLCI that is carrying the designated traffic type. A DLCI can be subdivided into 255 subchannels.
cid-type	The type of traffic carried on this subchannel.
input-pkts	Number of packets received by this subchannel.
output-pkts	Number of packets sent on this subchannel.
dropped-pkts	Total number of packets discarded by this subchannel.

Related Commands

Command	Description
show call active voice	Displays the contents of the active call table.
show call history voice	Displays the contents of the call history table.
show dial-peer voice	Displays configuration information and call statistics for dial peers.
show frame-relay fragment	Displays Frame Relay fragmentation details.
show frame-relay pvc	Displays statistics about PVCs for Frame Relay interfaces.
show voice-port	Displays configuration information about a specific voice port.

show gatekeeper calls

To show the status of each ongoing call that a gatekeeper is aware of, use the **show gatekeeper calls** command in privileged EXEC mode.

show gatekeeper calls

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.
12.0(5)T	The output for this command was changed.

Usage Guidelines

Use the **show gatekeeper calls** command to show all active calls currently being handled by a particular MCM gatekeeper. If you have forced a disconnect for either a particular call or all calls associated with a particular MCM gatekeeper by using the **clear h323 gatekeeper call** command, the system will not display information about those calls.

Examples

The following is sample output from the **show gatekeeper calls** command:

```
router# show gatekeeper calls
Total number of active calls = 1.
                         GATEKEEPER CALL INFO
                         ====================
LocalCallID                          Age(secs)    BW
12-3339                              94           768(Kbps)
 Endpt(s):Alias          E.164Addr     CallSignalAddr  Port  RASSignalAddr   Port
   src EP:epA                          90.0.0.11       1720  90.0.0.11       1700
   dst EP:epB@zoneB.com
   src PX:pxA                          90.0.0.01       1720  90.0.0.01       24999
   dst PX:pxB                          172.21.139.90   1720  172.21.139.90   24999
```

Table 40 describes the fields contained in the **show gatekeeper calls** sample output.

Table 40 show gatekeeper calls Field Descriptions

Field	Description
LocalCallID	Identification number of the call.
Age(secs)	The age of the call in seconds.

Table 40 show gatekeeper calls Field Descriptions (continued)

Field	Description
BW(Kbps)	The bandwidth in use in kilobits per second.
Endpoint(s)	Lists the role of each endpoints (terminal, gateway, or proxy) in the call (originator, target, or proxy), and the call signalling and RAS address.
Alias	H.323-ID or Email-ID of the endpoint.
E.164Addr	E.164 address of the endpoint.
CallSignalAddr	Call signalling IP address of the endpoint.
Port	Call signalling port number of the endpoint.
RASSignalAddr	RAS IP address of the endpoint.
Port	RAS port number of the endpoint.

Related Commands

Command	Description
clear h323 gateway call	Forces a specific call or all calls currently active on the gatekeeper to disconnect.

show gatekeeper endpoints

To display the status of all registered endpoints for a gatekeeper, use the **show gatekeeper endpoints** command in EXEC mode.

show gatekeeper endpoints

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

EXEC

Command History

Release	Modification
11.3(2)NA	This command was introduced.
12.0(5)T	The display format was modified for H.323 Version 2.

Usage Guidelines

Use this command to display the status of all registered endpoints for a gatekeeper.

Examples

The following is sample output from the **show gatekeeper endpoints** command:

```
Router# show gatekeeper endpoints
CallsignalAddr   Port  RASSignalAddr    Port   Zone Name    Type     F
---------------  ----  -------------    -----  ----------   -----    --
172.21.127.8     1720  172.21.127.8     24999  sj-gk        MCU
           H323-ID:joe@cisco.com
172.21.13.88     1720  172.21.13.88     1719   sj-gk        VOIP-GW   O     H323-ID:la-gw
```

Table 41 describes the fields contained in the **show gatekeeper endpoints** sample output.

Table 41 show gatekeeper endpoints Field Descriptions

Field	Description
CallsignalAddr	Call signalling IP address of the endpoint. If the endpoint also registered with alias(s), a list of all aliases registered for that endpoint should also be listed on the line below.
Port	Call signalling port number of the endpoint.
RASSignalAddr	RAS IP address of the endpoint.
Port	RAS port number of the endpoint.
Zone Name	Zone name (gatekeeper ID) that this endpoint registered in.

Table 41 show gatekeeper endpoints Field Descriptions (continued)

Field	Description
Type	The endpoint type (for example, terminal, gateway, or MCU).
F	S—Indicates that the endpoint is statically entered from the **alias** command—rather than dynamically registered through RAS messages. O—Indicates that the endpoint, which is a gateway, has sent notification that it is almost out of resources.

Related Commands

Command	Description
show gatekeeper gw-type-prefix	Displays the gateway technology prefix table.
show gatekeeper zone status	Displays the status of zones related to a gatekeeper.
show gateway	Displays the current gateway status.

show gatekeeper gw-type-prefix

To display the gateway technology prefix table, use the **show gatekeeper gw-type-prefix** command in privileged EXEC mode.

show gatekeeper gw-type-prefix

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(2)NA	This command was introduced.
12.0(5)T	The display format was modified for H.323 Version 2.

Usage Guidelines

Use the **show gatekeeper gw-type-prefix** command to display the gateway technology prefix table.

Examples

The following is sample output for a gatekeeper that is controlling two local zones, sj-gk and la-gk:

```
router# show gatekeeper gw-type-prefix

GATEWAY TYPE PREFIX TABLE
=========================
Prefix:12#*     (Default gateway-technology)
  Zone sj-gk master gateway list:
    172.21.13.11:1720 sj-gw1
    172.21.13.22:1720 sj-gw2 (out-of-resources)
    172.21.13.33:1720 sj-gw3
  Zone sj-gk prefix 408....... priority gateway list(s):
   Priority 10:
    172.21.13.11:1720 sj-gw1
   Priority 5:
    172.21.13.22:1720 sj-gw2 (out-of-resources)
    172.21.13.33:1720 sj-gw3
Prefix:7#*     (Hopoff zone la-gk)
  Statically-configured gateways (not necessarily currently registered):
    1.1.1.1:1720
    2.2.2.2:1720
  Zone la-gk master gateway list:
    171.69.127.11:1720 la-gw1
    171.69.127.22:1720 la-gw2
```

Table 42 describes the fields contained in the **show gatekeeper gw-type-prefix** sample output.

Table 42 show gatekeeper gw-type-prefix Field Descriptions

Field	Description
Prefix	The technology prefix defined with the **gw-type-prefix** command.
Zone sj-gk master gateway list	A list of all the gateways registered to zone sj-gk with the technology prefix, under which they are listed. (This display shows that gateways sj-gw1, sj-gw2, and sj-gw3 have registered in zone sj-gk with the technology prefix 12#.)
Zone sj-gk prefix 408....... priority gateway list(s)	A list of prioritized gateways to handle calls to area code 408.
Priority 10	Highest priority level. Gateways listed under priority 10 are given the highest priority when selecting a gateway to service calls to the specified area code. (In this display, gateway sj-gw1 is given the highest priority to handle calls to the 408 area code.)
Priority 5	Any gateway that does not have a priority level assigned to it defaults to priority 5.
(out-of-resources)	This is an indication that the displayed gateway has sent a "low-in-resources" notification.
(Hopoff zone la-gk)	Any call specifying this technology prefix should be directed to hop off in the la-gk zone, no matter what the area code of the called number is. (In this display, calls specifying technology prefix 7# are always routed to zone la-gk, regardless of the actual zone prefix in the destination address.)
Zone la-gk master gateway list	A list of all the gateways registered to la-gk with the technology prefix under which they are listed. (This display shows that gateways la-gw1 and la-gw2 have registered in zone la-gk with the technology prefix 7#. No priority lists are displayed here because none were defined for zone la-gk.)
(Default gateway-technology)	If no gateway-type prefix is specified in a called number, then gateways registering with 12# are the default type to be used for the call.
(Statically-configured gateways)	Lists all IP addresses and port numbers of gateways that are incapable of supplying technology-prefix information when they register. This display shows that when gateways 1.1.1.1:1720 and 2.2.2.2:1720 register, they will be considered to be of type 7#.

Related Commands

Command	Description
show gatekeeper calls	Displays the status of each ongoing call that a gatekeeper is aware of.

Command	Description
show gatekeeper endpoints	Displays the status of all registered endpoints for a gatekeeper.
show gateway	Displays the current gateway status.

show gatekeeper status

To show overall gatekeeper status that includes authorization and authentication status, zone status, and so on, use the **show gatekeeper status** command in EXEC mode.

show gatekeeper status

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

EXEC

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Examples

The following is sample output from the **show gatekeeper status** command:

```
router# show gatekeeper status

Gatekeeper State: UP
Zone Name: gk-px4.cisco.com
Accounting: DISABLED
Security: DISABLED
```

Table 43 describes the fields contained in the **show gatekeeper status** sample output.

Table 43 show gatekeeper status Field Descriptions

Field	Description
Gatekeeper State	The gatekeeper status: • UP is operational • DOWN is administratively shut down • INACTIVE is administratively enabled, that is, the no shutdown command has been issued but no local zones have been configured • HSRP STANDBY indicates the gatekeeper is on hot standby and will take over when the currently active gatekeeper fails.
Zone Name	Zone name.
Accounting	Authorization and accounting status.
Security	Security status.

show gatekeeper zone prefix

To display the zone prefix table, use the **show gatekeeper zone prefix** command in EXEC mode.

show gatekeeper zone prefix

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Examples

The following is an example from the **show gatekeeper zone prefix** command:

```
5300# show gatekeeper zone prefix

      ZONE PREFIX TABLE
      =================
GK-NAME                E164-PREFIX
-------                -----------
gk.zone13              212.......
gk.zone14              415.......
  gk.zone14          408.......
```

Table 44 describes the fields shown in the **show gatekeeper zone prefix** display.

Table 44 show gatekeeper zone prefix command Field Descriptions

Field	Description
GK-NAME	The gatekeeper name.
E164-PREFIX	The E.164 prefix and a dot that acts as a wildcard for matching each remaining number in the telephone number.

show gatekeeper zone status

To display the status of zones related to a gatekeeper, use the **show gatekeeper zone status** command in privileged EXEC mode.

show gatekeeper zone status

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(2)NA	This command was introduced.
12.0(5)T	This display format was modified for H.323 Version 2.

Usage Guidelines

Use this command to display the status of all zones related to a gatekeeper.

Examples

The following is an example from the **show gatekeeper zone status** command:

```
router# show gatekeeper zone status

                      GATEKEEPER ZONES
                      ================
GK name       Domain Name   RAS Address     PORT  FLAGS MAX-BW    CUR-BW
                                                        (kbps)    (kbps)
-------       -----------   -----------     ----  ----- ------    ------
sj.xyz.com    xyz.com       1.14.93.85      1719  LS              0
  SUBNET ATTRIBUTES :
    All Other Subnets :(Enabled)
  PROXY USAGE CONFIGURATION :
    inbound Calls from germany.xyz.com :
      to terminals in local zone sj.xyz.com :use proxy
      to gateways in local zone sj.xyz.com  :do not use proxy
    Outbound Calls to germany.xyz.com
      from terminals in local zone germany.xyz.com :use proxy
      from gateways in local zone germany.xyz.com  :do not use proxy
    Inbound Calls from all other zones :
      to terminals in local zone sj.xyz.com :use proxy
      to gateways in local zone sj.xyz.com  :do not use proxy
    Outbound Calls to all other zones :
      from terminals in local zone sj.xyz.com :do not use proxy
      from gateways in local zone sj.xyz.com  :do not use proxy
tokyo.xyz.co xyz.com       172.21.139.89      1719  RS              0
milan.xyz.co xyz.com       171.69.57.90       1719  RS              0
```

Table 45 describes the fields contained in the **show gatekeeper zone status** sample output.

Table 45 show gatekeeper zone status Field Descriptions

Field	Description
GK name	The gatekeeper name (also known as zone name), which is truncated after 12 characters in the display.
Domain Name	The domain with which the gatekeeper is associated.
RAS Address	The RAS address of the gatekeeper.
FLAGS	Displays the following information: • S = Static (CLI-configured, not DNS-discovered) • L = Local • R = Remote
MAX-BW	The maximum bandwidth for the zone in kilobits per second.
CUR-BW	The current bandwidth in use, in kbps.
SUBNET ATTRIBUTES	A list of subnets controlled by the local gatekeeper.
PROXY USAGE CONFIGURATION	Inbound and outbound proxy policies as configured for the local gatekeeper (or zone).

Related Commands

Command	Description
show gatekeeper calls	Shows the status of each ongoing call that a gatekeeper is aware of.
show gatekeeper endpoints	Displays the status of all registered endpoints for a gatekeeper.
show gateway	Displays the current gateway status.

show gateway

To display the current gateway status, use the **show gateway** command in privileged EXEC mode.

show gateway

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(6)NA2	This command was introduced.
12.0(5)T	This display format was modified for H.323 V2.

Usage Guidelines

This command displays the current gateway status.

Examples

The following example shows the report that appears when the gateway is not registered with a gatekeeper:

```
gateway1# show gateway
Gateway gateway1 is not registered to any gatekeeper
Gateway alias list
H323-ID gateway1
H323 resource thresholding is Enabled but NOT Active
H323 resource threshold values:
    DSP: Low threshold 60, High threshold 70
    DS0: Low threshold 60, High threshold 70
```

This following example indicates that an E.164 address has been assigned to the gateway:

```
gateway1# show gate
 Gateway  gateway1 is registered to Gatekeeper gk1
 Gateway alias list
 E.164 Number 5551212
 H323-ID gateway1
```

The following example shows the report that appears when the gateway is registered with a gatekeeper and H.323 resource threshold reporting is enabled with the **resource threshold** command:

```
gateway1# show gateway
Gateway gateway1 is registered to Gatekeeper gk1
Gateway alias list
H323-ID gateway1
H323 resource thresholding is Enabled and Active
H323 resource threshold values:
    DSP: Low threshold 60, High threshold 70
    DS0: Low threshold 60, High threshold 70
```

The following example shows the report that appears when the gateway is registered with a gatekeeper and H.323 resource threshold reporting is disabled with the **no resource threshold** command:

```
gateway1# show gateway
Gateway gateway1 is registered to Gatekeeper gk1
Gateway alias list
H323-ID gateway1
H323 resource thresholding is Disabled
```

Related Commands

Command	Description
resource threshold	Configures a gateway to report H.323 resource availability to the gatekeeper of the gateway.

show interface cable

To display cable interface information, use the **show interface cable** command in privileged EXEC mode.

show interface cable *slot/port* [**downstream** | **upstream**]

Syntax Description

slot/port	Identifies the Cisco uBR7200 chassis slot number and downstream port number. Valid values are from 3 to 6.
downstream	(Optional) Displays cable downstream port information for a cable modem.
upstream	(Optional) Displays cable upstream port information for a cable modem.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 XA	This command was introduced.

Examples

The following is sample output from the **show interface cable** command for the cable modem card located in slot 6/port 0:

```
CMTS01# show interface cable 6/0
Cable6/0 is up, line protocol is up
  Hardware is BCM3210 FPGA, address is 00e0.1e5f.7a60 (bia 00e0.1e5f.7a60)
  Internet address is 1.1.1.3/24
  MTU 1500 bytes, BW 27000 Kbit, DLY 1000 usec, rely 255/255, load 1/255
  Encapsulation, loopback not set, keepalive not set
  ARP type: ARPA, ARP Timeout 04:00:00
  Last input 4d07h, output 00:00:00, output hang never
  Last clearing of "show interface" counters never
  Queueing strategy: fifo
  Output queue 0/40, 0 drops; input queue 0/75, 0 drops
  5 minute input rate 0 bits/sec, 0 packets/sec
  5 minute output rate 0 bits/sec, 0 packets/sec
     10908 packets input, 855000 bytes, 0 no buffer
     Received 3699 broadcasts, 0 runts, 0 giants, 0 throttles
     3 input errors, 3 CRC, 0 frame, 0 overrun, 0 ignored, 0 abort
     5412 packets output, 646488 bytes, 0 underruns
     0 output errors, 0 collisions, 13082 interface resets
     0 output buffer failures, 0 output buffers swapped out
```

Table 46 describes the fields shown in the **show interface cable** display.

Table 46 show interface cable Field Descriptions

Field	Description
Cable slot/port is up/...administratively down	Indicates whether the interface hardware is currently active or taken down by the administrator.
line protocol is up/...administratively down	Indicates whether the software processes that handle the line protocol believe the interface is usable or if it has been taken down by the administrator.
hardware	Hardware type and address.
Internet address	Internet address followed by subnet mask.
MTU	Maximum Transmission Unit (MTU) of the interface.
BW	Bandwidth of the interface in kilobits per second.
DLY	Delay of the interface in microseconds.
rely	Reliability of the interface as a fraction of 255, calculated as an exponential average over 5 minutes. (For example, 255/255 is 100% reliability.)
load	Load on the interface as a fraction of 255, calculated as an exponential average over 5 minutes. (For example, 255/255 is complete saturation.)
Encapsulation	Encapsulation method assigned to this interface.
ARP type	Type of Address Resolution Protocol (ARP) and timeout value assigned.
Last input	Number of hours, minutes, and seconds since the last packet was successfully received by an interface.
output	Number of hours, minutes, and seconds since the last packet was successfully sent by an interface.
Last clearing of "show interface" counters	Time at which the counters that measure cumulative statistics (such as number of bytes sent and received) were last reset to zero.
Queueing strategy	Displays the type of queueing configured for this interface. In the following example output, the type of queueing configured is First In First Out (FIFO).
Output queue	Number of packets in the output queue. The format of this number is A/B, where A indicates the number of packets in the queue, and B indicates the maximum number of packets allowed in the queue.
drops	Indicates the number of packets dropped due to a full queue.
input queue/drops	Number of packets in the input queue. The format of this number is A/B, where A indicates the number of packets in the queue, and B indicates the maximum number of packets allowed in the queue.
drops	Indicates the number of packets dropped due to a full queue.
Five minute input rate Five minute output rate	Average number of bits and packets sent per second in the last five minutes.
packets input	Total number of error-free packets received by the system.

Table 46 show interface cable Field Descriptions (continued)

Field	Description
bytes input	Total number of bytes, including data and MAC encapsulation, in the error-free packets received by the system.
no buffer	Number of received packets discarded because there was no buffer space in the main system.
Received broadcast	Total number of broadcast or multicast packets received by the interface.
runts	Number of packets that are discarded because they are smaller than the medium's minimum packet size.
giants	Number of packets that are discarded because they exceed the medium's maximum packet size.
input errors	Includes runts, giants, no buffers, CRC, frame, overrun, and ignored counts.
CRC	Indicates the number of times the cyclic redundancy checksum generated by the originating LAN station or far-end device does not match the checksum calculated from the data received.
frame	Number of packets received incorrectly having a CRC error and a non-integer number of octets.
overrun	Number of times the receiver hardware was unable to forward received data to a hardware buffer because the input rate exceeded the receiver's ability to handle the data.
ignored	Number of received packets ignored by the interface because the interface hardware ran low on internal buffers.
packets output	Total number of messages sent by the system.
bytes	Total number of bytes, including data and MAC encapsulation, sent by the system.
underruns	Number of times the sender has been running faster than the receiving device can handle.
output errors	Sum of all errors that prevented the final transmission of packets out of the interface being examined.
collisions	Not applicable to the Cisco uBR7246.
interface resets	Number of times an interface has been completely reset.
output buffer failures	Number of times the output buffer has failed.
output buffer swapped out	Number of times the output buffer has been swapped out.

The following is sample output from the **show interface cable downstream** command for the downstream cable interface of slot 6 on port 0:

```
CMTS01# show interface cable 6/0 downstream
Cable6/0: Downstream is up
     111947771 packets output, 1579682655 bytes, 0 discarded
     0 output errors
```

Table 47 describes the fields shown in the **show interface cable downstream** display.

Table 47 show interface cable downstream Field Descriptions

Field	Description
Cable	Indicates the location of the downstream interface.
Downstream is up/...administratively down	Indicates the administrative state of the interface.
packets output	Total number of packets sent out of this interface.
bytes	Total number of bytes sent out of this interface.
discarded	Total number of packets discarded.
output errors	Sum of all errors that prevented downstream transmission of packets out of this interface.

The following is sample output for the upstream cable interface located in slot 6/port 0 from the **show interface cable upstream** command:

```
CMTS01# show interface cable 6/0 upstream
Cable6/0: Upstream 0 is up
Received 3699 broadcasts, 0 multicasts, 28586 unicasts
0 discards, 0 errors, 0 unknown protocol
21817 packets input, 0 corrected, 0 uncorrectable
0 noise, 0 microreflections
Guaranteed-rate service queue depth:0
Best-effort service queue depth:0
Total Modems On This Upstream Channel:3 (3 active)
Current Total Bandwidth Reserved:192000 bps
Current Admission Control Status: ENFORCED
Percentage of Oversubscription: 200%
Reservation Limit (with Oversubscription):5120000 bps
Last Minislot Stamp (current_time_base):190026    FLAG:1
Last Minislot Stamp (scheduler_time_base):200706  FLAG:1
```

Table 48 describes the fields shown in the **show interface cable upstream** display.

Table 48 show interface cable upstream Field Descriptions

Field	Description
Cable	Indicates the location of the upstream interface.
Upstream is up/...administratively down	Indicates the administrative state of the upstream interface.
Received broadcasts	Number of broadcast packets received through this upstream interface.
multicasts	Number of multicast packets received through this upstream interface.
unicasts	Number of unicast packets received through this interface.
discards	Number of packets discarded by this interface.
errors	Sum of all errors that prevented upstream transmission of packets through this interface.

Table 48 show interface cable upstream Field Descriptions (continued)

Field	Description
unknown protocol	Number of packets received that were generated using a protocol unknown to the Cisco uBR7246.
packets input	Number of packets received through this upstream interface that were free from errors.
corrected	Number of error packets received through this upstream interface that were corrected.
uncorrectable	Number of error packets received through this upstream interface that could not be corrected.
noise	Number of upstream packets corrupted by line noise.
microreflections	Number of upstream packets corrupted by microreflections.
Guaranteed-rate service queue depth	Number of bandwidth requests queued up in the Guarantee-rate queue. This queue is only available to modems that have a reserved minimum upstream rate in their Class of Service.
Best-effort service queue depth	Number of bandwidth requests queued up in the Best-effort queue. This queue is available to all modems that do not have any reserved rate on the upstream.
Total Modems On This Upstream Channel	Number of cable modems currently sharing this upstream channel. This field also shows how many of these modems are active.
Current Total Bandwidth Reserved	Total amount of bandwidth reserved by all modems sharing this upstream channel that require bandwidth reservation. The Class of Service for these modems specifies some non-zero value for the guaranteed-upstream rate. When one of these modems is admitted on the upstream, this field value is incremented by this guaranteed-upstream rate value.
Current Admission Control Status	Indicates the status of admission control on the upstream channel. ENFORCED status allows users to enable admission control on a per port basis. This controls how limited bandwidth is allocated. NOT ENFORCED status indicates that there is no admission control. Every modem that registers with a class of service specifying a minimum upstream rate will be admitted by the CMTS regardless of how much aggregate bandwidth is actually available. Users enable admission control via the admission control CLI.
Percentage of Oversubscription	Amount of oversubscription to allow on this upstream channel. Oversubscription is expressed as a percentage of the raw capacity of the channel. In the example shown, an oversubscription rate of 200% on a 2.56 Mbps channel allows the cumulative bandwidth reservation on this channel to reach 5.12 Mbps before modems configured with non-zero reserved upstream rates are denied service.
Reservation Limit (with Oversubscription)	Maximum cumulative bandwidth reservation allowable before rejecting new modems. In the example shown, this reservation limit with oversubscription is 5.12 Mbps.

Table 48 show interface cable upstream Field Descriptions (continued)

Field	Description
Last Minislot Stamp (current_time_base)	Indicates the current minislot count at the CMTS. FLAG indicates the timebase reference. This field is used only by developers.
Last Minislot Stamp (scheduler_time_base)	Indicates the furthest minislot count allocated at the indicated time. FLAG indicates the timebase reference. This field is used by developers.

Related Commands

Command	Description
show interface cable sid	Displays information by SID of each cable modem on the network.
show interface cable signal-quality	Displays information about the cable signal quality.

show interface cable sid

To display information by service identifier (SID) of each cable modem on the HFC network, use the **show interface cable sid** command in privileged EXEC mode.

show interface cable *slot/port* **sid** [*sid-number*]l

Syntax Description

slot/port	Identifies the Cisco uBR7200 chassis slot number and downstream port number. Valid values are from 3 to 6.
sid-number	(Optional) Identifies the service identification number.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 XA	This command was introduced.
12.0(5)T	The command output was modified to identify secondary SIDs.

Usage Guidelines

Data transport over the RF link uses the registered SID address rather than the Ethernet address. This allows multiple hosts to access the network via a single cable modem.

Examples

The following example confirms that cable modem 0010.7b6b.7219 had two SIDs. The primary SID was 3, and the secondary SID was 8.

```
cmts# show interface cable 3/0 sid
 Sid  Prim Online     Admin   QoS Create       IP Address        MAC Address
      Sid  State      Status      Time
 1         online     enable  4   17:00:38     19.2.20.141       0010.7b6b.71cd
 2         online     enable  4   17:00:38     19.2.20.139       0010.7b6b.7215
 3         online     enable  5   17:00:40     19.2.20.145       0010.7b6b.7219
 8    3               enable  6   17:31:10
```

Table 49 describes the fields shown in the show interface cable display.

Table 49 show interface cable sid Field Descriptions

Field	Description
Sid	The secondary service ID assigned to the modem.
Prim Sid	The primary service ID assigned to the modem.
Admin Status	The status of the cable modem.
QoS	The service class assigned to the modem.

Table 49 show interface cable sid Field Descriptions (continued)

Field	Description
Create Time	When the SID was created, number of hours, minutes, and seconds since system booted.
IP address	IP address of the modem.
MAC address	Media access layer address of the modem.

Related Commands

Command	Description
show interface cable signal-quality	Displays information about the cable signal quality.

show interface cable signal-quality

To display information about the signal quality of a downstream port on a cable modem card in a Cisco uBR7200 series cable router, use the **show interface cable signal-quality** command in privileged EXEC mode.

show interface cable *slot/port* **signal-quality**

Syntax Description

slot/port	Identifies the Cisco uBR7200 chassis slot number and downstream port number.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 XA	This command was introduced.

Examples

The following is sample output from the **show interface cable signal-quality** command:

```
CMTS01# show interface cable 6/0 signal-quality
Cable6/0: Upstream 0 is up includes contention intervals: TRUE
```

Table 50 describes the fields shown in the **show interface cable signal-quality** display.

Table 50 show interface cable signal-quality Field Descriptions

Field	Description
Cable	Interface name.
Upstream is up includes contention intervals	States whether this statement is true.

Related Commands

Command	Description
show interface cable	Displays cable interface information.
show interface cable sid	Displays information by SID of each cable access router on the network.

show interfaces cable-modem

To display information about the Cisco uBR924 cable access router cable interface, use the **show interfaces cable-modem** command in either user EXEC mode or privileged EXEC mode.

show interfaces cable-modem *number* [**accounting** | **counters** | **crb** | **irb** | **type**]

Syntax Description

number	Cable access router interface number.
accounting	(Optional) Displays the number of packets of each protocol type that has been sent through the cable access router interface.
counters	(Optional) Shows MIB counters on the cable interface.
crb	(Optional) Displays concurrent routing and bridging information for each interface that has been configured for routing or bridging. This option does not really apply to the Cisco uBR924; it is included because it is part of the subsystem that provides DOCSIS-compliant bridging. For more information, refer to the *Bridging and IBM Networking Command Reference.*
irb	(Optional) Displays integrated routing and bridging information for each interface that has been configured for routing or bridging. This option does not really apply to the Cisco uBR924; it is included because it is part of the subsystem that provides DOCSIS-compliant bridging. For more information, refer to the *Bridging and IBM Networking Command Reference.*
type	(Optional) Designed to display information about virtual LANs associated with the interface; however, this option is not supported on the Cisco uBR924.

Defaults

No default behavior or values.

Command Modes

User EXEC or privileged EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

When this command is entered without a keyword, general information about the cable interface is displayed.

Examples

Traffic passing through the cable access router interface is shown in the following example:

```
uBR924# show interfaces cable-modem 0
  cable-modem0 is up, line protocol is up
  Hardware is BCM3300, address is 0050.7366.2439 (bia 0050.7366.2439)
  Internet address is 5.2.0.11/16
  MTU 1500 bytes, BW 27000 Kbit, DLY 1000 usec,
     reliability 255/255, txload 1/255, rxload 1/255
  Encapsulation DOCSIS, loopback not set
  Keepalive set (10 sec)
  ARP type:ARPA, ARP Timeout 04:00:00
  Last input 00:00:00, output 00:00:00, output hang never
  Last clearing of "show interface" counters 00:08:40
  Queueing strategy:fifo
  Output queue 40/40, 52787 drops; input queue 0/75, 0 drops
  5 minute input rate 2000 bits/sec, 2 packets/sec
  5 minute output rate 94000 bits/sec, 154 packets/sec
     1074 packets input, 418472 bytes, 0 no buffer
     Received 19 broadcasts, 0 runts, 0 giants, 0 throttles
     0 input errors, 0 CRC, 0 frame, 0 overrun, 0 ignored, 0 abort
     78771 packets output, 6326786 bytes, 0 underruns
     0 output errors, 0 collisions, 0 interface resets
     0 output buffer failures, 0 output buffers swapped out
```

Table 51 describes the significant fields shown in the display.

Table 51 show interfaces cable-modem Field Descriptions

Field	Description
cable-modem0 is up	Indicates that the interface is currently active. "Disabled" indicates the interface has received more than 5000 errors in one keepalive interval (10 seconds by default if keepalive is set); "administratively down" indicates the interface has been taken down by an administrator.
line protocol is up	Indicates that the software processes that handle the line protocol believe the interface is usable.
Hardware	Hardware type and MAC address.
Internet address	Internet address followed by the shorthand notation for the subnet mask.
MTU	Maximum Transmission Unit (equivalent of the maximum packet size) for the interface.
BW	Bandwidth of the interface in kilobits per second.
DLY	Delay of the interface in microseconds.
reliability	Reliability of the interface, expressed as a fraction of 255, calculated as an exponential average over a 5-minute period. (255/255 equals 100% reliability.)
tx load/rx load	Load on the interface caused by transmitting and receiving, expressed as a fraction of 255, calculated as an exponential average over a 5 minute period.
Encapsulation/loopback/keepalive	Encapsulation method assigned to the interface.
loopback	Indicates whether or not loopback is set.
keepalive	Indicates whether or not keepalives are set.

Table 51 show interfaces cable-modem Field Descriptions (continued)

Field	Description
ARP type	Type of Address Resolution Protocol configured for the interface.
ARP Timeout	Number of hours, minutes, and seconds an ARP cache entry will stay in the cache.
Last input/output	Number of hours, minutes, and seconds since the last packet was successfully received/transmitted by the interface.
output hang	Number of hours, minutes, and seconds since the interface was last reset because of a transmission that took too long. When the number of hours in any of the "Last..." fields exceeds 24, the number of days and hours is displayed. If the field overflows, asterisks are printed.
Last clearing of "show interface" counters	Time at which the counters that measure cumulative statistics (such as number of bytes transmitted and received) shown in this report were last reset to zero. Note that variables that might affect routing (for example, load and reliability) are not cleared when the counters are cleared. *** indicates the elapsed time is too large to be displayed. 0:00:00 indicates the counters were cleared more than 2^{31} milliseconds (and less than 2^{32} milliseconds) ago.
Queueing strategy	Type of queueing strategy in effect on the interface.
Output queue/drops	Number of packets in the output queue followed by the size of the queue and the number of packets dropped due to a full queue.
input queue/drops	Number of packets in the input queue followed by the size of the queue and the number of packets dropped due to a full queue.
5 minute input rate 5 minute output rate	Average number of bits and packets received and transmitted per second in the last 5 minutes. If the interface is not in promiscuous mode, it senses network traffic it sends and receives (rather than all network traffic). The 5-minute input and output rates should be used only as an approximation of traffic per second during a given 5-minute period. These rates are exponentially weighted averages with a time constant of 5 minutes. A period of four time constants must pass before the average will be within two percent of the instantaneous rate of a uniform stream of traffic over that period.
packets input	Total number of error-free packets received by the system.
bytes input	Total number of bytes, including data and MAC encapsulation, in the error-free packets received by the system.
no buffer	Number of received packets discarded because there was no buffer space in the main system. Compare with ignored count. Broadcast storms on Ethernet networks and bursts of noise on serial lines are often responsible for no input buffer events.

Table 51 show interfaces cable-modem Field Descriptions (continued)

Field	Description
Received broadcasts	Total number of broadcast or multicast packets received by the interface.
runts	Number of packets discarded because they were smaller than the medium's minimum packet size. For example, any Ethernet packet less than 64 bytes is considered a runt.
giants	Number of packets discarded because they were larger than the medium's maximum packet size. For example, any Ethernet packet larger than 1518 bytes is considered a giant.
throttles	Number of times the receiver on the port was disabled, possibly due to buffer or processor overload.
input errors	Includes runts, giants, no buffer, CRC, frame, overrun, and ignored counts. Other input-related errors can also cause the input errors count to be increased, and some datagrams may have more than one error; therefore, this sum may not balance with the sum of enumerated input error counts.
CRC	Number of cyclic redundancy checksums generated by the originating LAN station or far-end device that do not match the checksum calculated from the data received. On a LAN, this usually indicates noise or transmission problems on the LAN interface or the LAN bus itself. A high number of CRCs is usually the result of collisions or a station sending bad data.
frame	Number of packets received incorrectly, having a CRC error and a noninteger number of octets. On a LAN, this is usually the result of collisions or a malfunctioning Ethernet device.
overrun	Number of times the receiver hardware was unable to hand received data to a hardware buffer because the input rate exceeded the receiver's ability to handle the data.
ignored	Number of received packets ignored by the interface because the interface hardware ran low on internal buffers. These buffers are different from the system buffers mentioned previously in the buffer description. Broadcast storms and bursts of noise can cause the ignored count to be increased.
abort	Number of packets whose receipt was aborted.
packets output	Total number of messages sent by the system.
bytes	Total number of bytes, including data and MAC encapsulation, sent by the system.
underruns	Number of times the transmitter has been running faster than the router can handle.
output errors	Sum of all errors that prevented the final transmission of datagrams out of the interface being examined. Note that this may not balance with the sum of the enumerated output errors, as some datagrams might have more than one error, and others might have errors that do not fall into any of the specifically tabulated categories.

Table 51 show interfaces cable-modem Field Descriptions (continued)

Field	Description
collisions	Number of messages retransmitted due to an Ethernet collision. This is usually the result of an overextended LAN (Ethernet or transceiver cable too long, more than two repeaters between stations, or too many cascaded multiport transceivers). A packet that collides is counted only once in output packets.
interface resets	Number of times an interface has been completely reset. This can happen if packets queued for transmission were not sent within several seconds. On a serial line, this can be caused by a malfunctioning modem that is not supplying the transmit clock signal, or by a cable problem. If the system notices that the carrier detect line of a serial interface is up, but the line protocol is down, it periodically resets the interface in an effort to restart it. Interface resets can also occur when an interface is looped back or shut down.
output buffer failures	Number of times the output buffer has failed.
output buffers swapped out	Number of times the output buffer has been swapped out.

To display the number of packets and bytes of each protocol type passing through the cable access router interface, use the **accounting** option with the **show interfaces cable-modem** command:

```
uBR924# show interfaces cable-modem 0 accounting
cable-modem0
                Protocol    Pkts In   Chars In   Pkts Out  Chars Out
                      IP        545     185502        159      90240
           Trans. Bridge       3878     964995      12597    1611142
                     ARP         73       3066         86       4128
```

Table 52 describes the significant fields shown in this display.

Table 52 show interfaces cable-modem accounting Field Descriptions

Field	Description
Protocol	List of protocols operating on the cable-modem interface.
Pkts In	Number of packets of each protocol received on the interface.
Chars In	Number of bytes of each protocol received on the interface.
Pkts Out	Number of packets of each protocol sent on the interface.
Chars Out	Number of bytes of cache protocol sent on the interface.

MIB counters on the cable interface are displayed in the following example:

```
uBR924# show interfaces cable-modem 0 counters
Cable specific counters:
Ranging requests sent  : 50982
Downstream FIFO full   : 0
Re-requests            : 7277
DS MAC Message Overruns: 0
DS Data Overruns       : 0
Received MAPs          : 254339485
Received Syncs         : 53059555
Message CRC failures   : 0
Header CRC failures    : 1394
Data PDUs              : 5853
DS MAC messages        : 307861745
Valid Headers          : 307869065
Sync losses            : 0
Pulse losses           : 1
BW request failures    : 6
```

Table 53 describes the counters shown in this display.

Table 53 Counters Shown in show interfaces cable-modem counters Display

Field	Description
Ranging requests sent	Number of ranging requests sent by the Cisco uBR924 to the CMTS.
Downstream FIFO full	Number of times the downstream input first-in first-out (FIFO) buffer became full on the Cisco uBR924.
Re-requests	Number of times a bandwidth request generated by the Cisco uBR924 was not responded to by the CMTS.
DS MAC Message Overruns	Number of times the Cisco uBR924 DMA controller had a downstream MAC message and there were no free MAC message buffer descriptors to accept the message.
DS Data Overruns	Number of times the Cisco uBR924 DMA controller had downstream data and there were no free data PDU buffer descriptors to accept the data.
Received MAPs	Number of times a MAP message passed all filtering requirements and was received by the Cisco uBR924.
Received Syncs	Number of times a time-stamp message was received by the Cisco uBR924.
Message CRC failures	Number of times a MAC message failed a cyclic redundancy (CRC) check.
Header CRC failures	Number of times a MAC header failed its 16-bit CRC check. The MAC header CRC is a 16-bit Header Check Sequence (HCS) field that ensures the integrity of the MAC header even in a collision environment.
Data PDUs	Total number of data PDUs (protocol data units) of all types received by the Cisco uBR924.
DS MAC messages	Number of MAC messages received by the Cisco uBR924.
Valid Headers	Number of valid headers received by the Cisco uBR924, including PDU headers, MAC headers, and headers only.
Sync losses	Number of times the Cisco uBR924 lost timebase sync with the CMTS.

Table 53 Counters Shown in show interfaces cable-modem counters Display (continued)

Field	Description
Pulse losses	Number of times the Cisco uBR924 did not receive expected timestamp messages from the CMTS.
BW request failures	Number of times the Cisco uBR924 sent the maximum number of re-requests for bandwidth allocation and the request was still not granted.

Information about routing and bridging protocols and filtering on the cable access router interface is displayed in the following example:

```
uBR924# show interfaces cable-modem 0 crb

cable-modem0

 Bridged protocols on cable-modem0:
  ip

 Software MAC address filter on cable-modem0
  Hash Len    Address      Matches  Act      Type
  0x00:  0 ffff.ffff.ffff     3877 RCV Physical broadcast
  0x2A:  0 0900.2b01.0001        0 RCV DEC spanning tree
  0x7A:  0 0010.7b43.aa01      573 RCV Interface MAC address
  0xC2:  0 0180.c200.0000        0 RCV IEEE spanning tree
  0xC2:  1 0180.c200.0000        0 RCV IBM spanning tree
```

Table 54 describes the software MAC address filter information for the cable access router interface.

Table 54 Software MAC Address Filter Information

Field	Description
Hash	Hash key/relative position in the keyed list for this MAC address filter.
Len	Length of this entry to the beginning element of this hash chain.
Address	Canonical (Ethernet ordered) MAC address of this filter.
Matches	Number of received packets that match this MAC address.
Act	Action to be taken when this address is looked up; choices are to receive or discard the packet.
Type	MAC address type.

Related Commands

Command	Description
show bridge cable-modem	Displays bridging information for a cable modem.

show interface dspfarm dsp

To display digital signal processor (DSP) information on the two-port T1/E1 high-density port adapter for the Cisco 7200 series, use the **show interface dspfarm** command in privileged EXEC mode.

show interface dspfarm [*slot/port*] **dsp** [*number*] [**long** | **short**]

Syntax Description

slot	(Optional) Slot location of the port adapter.
port	(Optional) Port number on the port adapter.
number	(Optional) Specifies the number of DSP sets to show. The range is 1 to 30.
long	(Optional) Specifies detailed DSP information.
short	(Optional) Specifies brief DSP information.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)XE	This command was introduced.

Examples

The following example is sample output from the **show interface dspfarm** command in chassis slot 3, in port adapter slot 0 on the Cisco 7200 series router:

```
router# show interface dspfarm 3/0
DSPfarm3/0 is up, line protocol is up
  Hardware is VXC-2T1/E1
  MTU 256 bytes, BW 12000 Kbit, DLY 0 usec,
     reliability 255/255, txload 4/255, rxload 1/255
  Encapsulation VOICE, loopback not set
  C549 DSP Firmware Version:MajorRelease.MinorRelease (BuildNumber)
     DSP Boot Loader:255.255 (255)
     DSP Application:4.0 (3)
     Medium Complexity Application:3.2 (5)
     High Complexity Application:3.2 (5)
  Total DSPs 30, DSP0-DSP29, Jukebox DSP id 30
  Down DSPs:none
  Total sig channels 120 used 24, total voice channels 120 used 0
     0 active calls, 0 max active calls, 0 total calls
     30887 rx packets, 0 rx drops, 30921 tx packets, 0 tx frags
     0 curr_dsp_tx_queued, 29 max_dsp_tx_queued
  Last input never, output never, output hang never
  Last clearing of "show interface" counters never
  Queueing strategy:fifo
  Output queue 0/0, 0 drops; input queue 0/75, 0 drops
  5 minute input rate 13000 bits/sec, 94 packets/sec
  5 minute output rate 193000 bits/sec, 94 packets/sec
     30887 packets input, 616516 bytes, 0 no buffer
     Received 0 broadcasts, 0 runts, 0 giants, 0 throttles
```

```
0 input errors, 0 CRC, 0 frame, 0 overrun, 0 ignored, 0 abort
30921 packets output, 7868892 bytes, 0 underruns
0 output errors, 0 collisions, 0 interface resets
0 output buffer failures, 0 output buffers swapped out
```

Table 55 describes the fields contained in the show gatekeeper zone status sample output.

Table 55 show interface dspfarm Field Descriptions

Field	Description
DSPfarm3/0 is up	DSPfarm interface is operating. The interface state can be up, down, and administratively down.
Line protocol is	Indicates whether the software processes that handle the line protocol consider the line usable or if it has been taken down by an administrator.
Hardware	Version number of the hardware.
MTU	256 bytes.
BW	12000 Kbit.
DLY	Delay of the interface in microseconds.
Reliability	Reliability of the interface as a fraction of 255 (255/255 is 100% reliability, calculated as an expediential average over 5 minutes).
Txload	Number of packets sent.
Rxload	Number of packets received.
Encapsulation	Encapsulation method assigned to interface.
Loopback	Loopback conditions.
C549 DSP Firmware Version	The version of DSP firmware installed.
DSP Boot Loader	DSP boot loader version.
DSP Application	DSP application code version.
Medium Complexity Application	DSP Medium Complexity Application code version.
High Complexity Application	DSP High Complexity Application code version.
Total DSPs	Total DSPs that are equipped in the PA.
DSP0-DSP	DSP number range.
Jukebox DSP id	Jukebox DSP number.
Down DSPs	DSPs not in service.
Total sig channels...used...	Total number of signal channels used.
Total voice channels...used...	Total number of voice channels used.
Active calls	Number of active calls.
Max active calls	Maximum number of active calls.
Total calls	Total number of calls.
Rx packets	Number of received packets.
Rx drops	Number of rx packets dropped at PA.
Tx packets	Number of transmit packets.

Table 55 show interface dspfarm Field Descriptions (continued)

Field	Description
Tx frags	Number of tx packets that were fragmented.
Curr_dsp_tx_queued	Number of tx packets that are being queued at host DSP queues.
Max_dsp_tx_queued	The max total tx packets that were queued at host DSP queues.
Last input	Number of hours, minutes, and seconds since the last packet was successfully received by an interface. Useful for knowing when a dead interface failed.
Output	Number of hours, minutes, and seconds since the last packet was successfully sent by the interface. Useful for knowing when a dead interface failed.
Output hang	Number of hours, minutes, and seconds (or never) since the interface was last reset because of a transmission that took too long. When the number of hours in any of the "last" fields exceeds 24 hours, the number of days and hours is printed. If that field overflows, asterisks (**) are printed.
Last clearing of "show interface" counters	Number of times the "show interface" counters was cleared.
queueing strategy	First-in, first-out queueing strategy (other queueing strategies you might see are priority-list, custom-list, and weighted fair).
Output queue	Number of packets in output queue.
Drops	The number of packets dropped due to a full queue.
Input queue	Number of packets in input queue.
Minute input rate	Average number of bits and packets received per minute in the last 5 minutes.
Bits/sec	Average number of bits sent per second.
Packets/sec	Average number of packets sent per second.
Packets input	Total number of error-free packets received by the system.
Bytes	Total number of bytes, including data and MAC encapsulation, in the error free packets received by the system.
No buffer	Number of received packets discarded because there was not buffer space in the main system. Compare with ignored count. Broadcast storms on Ethernets and bursts of noise on serial lines are often responsible for no input buffer events.
Received...broadcasts	Total number of broadcast or multicast packets received by the interface.
Runts	Number of packets that are discarded because they are smaller than the medium's minimum packet size. For instance, any Ethernet packet that is less than 64 bytes is considered a runt.
Giants	Number of packets that are discarded because they exceed the medium's minimum packet size. For instance, any Ethernet packet that is greater than 1,518 bytes is considered a giant.
Throttles	Number of times the receiver on the port was disabled, possibly due to buffer or processor overload.

Table 55 show interface dspfarm Field Descriptions (continued)

Field	Description
Input errors	Number of packet input errors.
CRC	Cyclic redundancy checksum generated by the originating LAN station or far-end device does not match the checksum calculated from the data received. On a LAN, this usually indicates noise or transmission problems on the LAN interface or the LAN bus itself. A high number of CRCs is usually the result of collisions or a station sending bad data. On a serial link, CRCs usually indicate noise, gain hits or other transmission problems on the data link.
Frame	Number of packets received incorrectly having a CRC error and a non-integer number of octets. On a serial line, this is usually the result of noise or other transmission problems.
Overrun	Number of times the serial receiver hardware was unable to hand received data to a hardware buffer because the input rate exceeded the receiver's ability to handle the data.
Ignore	Number of received packets ignored by the interface because the interface hardware ran low on internal buffers. These buffers are different than the system buffers mentioned previously in the buffer description. Broadcast storms and bursts of noise can cause the ignored count to be incremented.
Abort	Illegal sequence of one bits on the interface.
Packets output	Total number of messages sent by the system.
Bytes	Total number of bytes, including data and MAC encapsulation, sent by the system.
Underruns	Number of times that the far-end transmitter has been running faster than the near-end router's receiver can handle.
Output errors	Sum of all errors that prevented the final transmission of datagrams out of the interface being examined. Note that this might not balance with the sum of the enumerated output errors, as some datagrams can have more than one error, and others can have errors that do not fall into any of the specifically tabulated categories.
Collisions	Number of messages resent due to an Ethernet collision. This is usually the result of an over extended LAN (Ethernet or transceiver cable too long, more than two repeaters between stations, or too many cascaded multiport transceivers). A packet that collides is counted only once in output packets.
Interface resets	Number of times an interface has been completely reset. This can happen if packets queued for transmission were not sent within a certain interval. If the system notices that the carrier detect line of an interface is up, but the line protocol is down, it periodically resets the interface in an effort to restart it. Interface resets can also occur when an unrecoverable interface processor error occurred, or when an interface is looped back or shut down.
Output buffer failures	Number of failed buffers.
Output buffers swapped out	Number of buffers swapped out.

show num-exp

To show the number expansions configured, use the **show num-exp** command in privileged EXEC mode.

show num-exp [*dialed-number*]

Syntax Description

dialed-number	(Optional) Dialed number.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **show num-exp** privileged EXEC command to display all of the number expansions configured for this router. To display number expansion for only one number, specify that number by using the *dialed-number* argument.

Examples

The following is sample output from the **show num-exp** command:

```
router# show num-exp
Dest Digit Pattern = '0...'     Translation = '+14085270...'
Dest Digit Pattern = '1...'     Translation = '+14085271...'
Dest Digit Pattern = '3..'      Translation = '+140852703..'
Dest Digit Pattern = '4..'      Translation = '+140852804..'
Dest Digit Pattern = '5..'      Translation = '+140852805..'
Dest Digit Pattern = '6....'    Translation = '+1408526....'
Dest Digit Pattern = '7....'    Translation = '+1408527....'
Dest Digit Pattern = '8...'     Translation = '+14085288...'
```

Table 56 explains the fields in the sample output.

Table 56 show num-exp Field Descriptions

Field	Description
Dest Digit Pattern	Index number identifying the destination telephone number digit pattern.
Translation	Expanded destination telephone number digit pattern.

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 pots status

To display the settings of the telephone port physical characteristics and other information on the telephone interfaces of the Cisco 800 series, use the **show pots status** command in privileged EXEC mode.

show pots status [**1** | **2**]

Syntax Description

1	(Optional) Display the settings of telephone port 1.
2	(Optional) Display the settings of telephone port 2.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

The **show pots status** command displays the settings and information for both telephone ports.

Examples

The following is a sample output from the **show pots status** command.

```
router # show pots status
POTS Global Configuration:
   Country: United States
   Dialing Method: Overlap, Tone Source: Remote, CallerId Support: YES
   Line Type: 600 ohm, PCM Encoding: u-law, Disc Type: OSI,
   Ringing Frequency: 20Hz, Distinctive Ring Guard timer: 0 msec
   Disconnect timer: 1000 msec, Disconnect Silence timer: 5 sec
   TX Gain: 6dB, RX Loss: -6dB,
   Filter Mask: 6F
   Adaptive Cntrl Mask: 0
POTS PORT: 1
   Hook Switch Finite State Machine:
      State: On Hook, Event: 0
      Hook Switch Register: 10, Suspend Poll: 0
   CODEC Finite State Machine:
      State: Idle, Event: 0
      Connection: None, Call Type: Two Party, Direction: Rx only
      Line Type: 600 ohm, PCM Encoding: u-law, Disc Type: OSI,
      Ringing Frequency: 20Hz, Distinctive Ring Guard timer: 0 msec
      Disconnect timer: 1000 msec, Disconnect Silence timer: 5 sec
      TX Gain: 6dB, RX Loss: -6dB,
      Filter Mask: 6F
      Adaptive Cntrl Mask: 0
```

```
      CODEC Registers:
         SPI Addr: 2, DSLAC Revision: 4
         SLIC Cmd: 0D, TX TS: 00, RX TS: 00
         Op Fn: 6F, Op Fn2: 00, Op Cond: 00
         AISN: 6D, ELT: B5, EPG: 32 52 00 00
         SLIC Pin Direction: 1F
      CODEC Coefficients:
         GX: A0 00
         GR: 3A A1
          Z: EA 23 2A 35 A5 9F C2 AD 3A AE 22 46 C2 F0
          B: 29 FA 8F 2A CB A9 23 92 2B 49 F5 37 1D 01
          X: AB 40 3B 9F A8 7E 22 97 36 A6 2A AE
          R: 01 11 01 90 01 90 01 90 01 90 01 90
         GZ: 60
        ADAPT B: 91 B2 8F 62 31
      CSM Finite State Machine:
         Call 0 - State: idle, Call Id: 0x0
                  Active: no
         Call 1 - State: idle, Call Id: 0x0
                  Active: no
         Call 2 - State: idle, Call Id: 0x0
                  Active: no
   POTS PORT: 2
      Hook Switch Finite State Machine:
         State: On Hook, Event: 0
         Hook Switch Register: 20, Suspend Poll: 0
      CODEC Finite State Machine:
         State: Idle, Event: 0
         Connection: None, Call Type: Two Party, Direction: Rx only
         Line Type: 600 ohm, PCM Encoding: u-law, Disc Type: OSI,
         Ringing Frequency: 20Hz, Distinctive Ring Guard timer: 0 msec
         Disconnect timer: 1000 msec, Disconnect Silence timer: 5 sec
         TX Gain: 6dB, RX Loss: -6dB,
         Filter Mask: 6F
         Adaptive Cntrl Mask: 0
      CODEC Registers:
         SPI Addr: 3, DSLAC Revision: 4
         SLIC Cmd: 0D, TX TS: 00, RX TS: 00
         Op Fn: 6F, Op Fn2: 00, Op Cond: 00
         AISN: 6D, ELT: B5, EPG: 32 52 00 00
         SLIC Pin Direction: 1F
      CODEC Coefficients:
         GX: A0 00
         GR: 3A A1
          Z: EA 23 2A 35 A5 9F C2 AD 3A AE 22 46 C2 F0
          B: 29 FA 8F 2A CB A9 23 92 2B 49 F5 37 1D 01
          X: AB 40 3B 9F A8 7E 22 97 36 A6 2A AE
          R: 01 11 01 90 01 90 01 90 01 90 01 90
         GZ: 60
        ADAPT B: 91 B2 8F 62 31
      CSM Finite State Machine:
         Call 0 - State: idle, Call Id: 0x0
                  Active: no
         Call 1 - State: idle, Call Id: 0x0
                  Active: no
         Call 2 - State: idle, Call Id: 0x0
                  Active: no
   Time Slot Control: 0
```

Table 57 explains the fields in the **show pots status** command sample output.

Table 57 show pots status Field Descriptions

Field	Descriptions
POTS Global Configuration	Displays the settings of the telephone port physical characteristic commands. Also displays the following: • TX GAIN—Current transmit gain of telephone ports. • RX LOSS—Current transmit loss of telephone ports. • Filter Mask—Value determines which filters are currently enabled or disabled in the telephone port hardware. • Adaptive Cntrl Mask—Value determines if telephone port adaptive line impedance hardware is enabled or disabled.
Hook Switch Finite State Machine	Device driver that tracks state of telephone port hook switch.
CODEC Finite State Machine	Device driver that controls telephone port codec hardware.
CODEC Registers	Register contents of telephone port codec hardware.
CODEC Coefficients	Codec coefficients selected by telephone port driver. Selected line type determines codec coefficients.
CSM Finite State Machine	State of call-switching module (CSM) software.
Time Slot Control	Register that determines if telephone port voice or data packets are sent to an ISDN B channel.

Related Commands

Command	Description
pots country	Configures telephones, fax machines, or modems connected to a Cisco 800 series router to use country-specific default settings for each physical characteristic.
pots dialing-method	Specifies how the Cisco 800 series router collects and sends digits dialed on your connected telephones, fax machines, or modems.
pots disconnect-supervision	Specifies how a Cisco 800 series router notifies the connected telephones, fax machines, or modems when the calling party has disconnected.
pots disconnect-time	Specifies the interval in which the disconnect method is applied if telephones, fax machines, or modems connected to a Cisco 800 series router fail to detect that a calling party has disconnected.
pots distinctive-ring-guard-time	Specifies a delay in which a telephone port can be rung after a previous call is disconnected (Cisco 800 series routers).
pots encoding	Specifies the PCM encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router.
pots line-type	Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router.

Command	Description
pots ringing-freq	Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring.
pots silence-time	Specifies the interval of silence after a calling party disconnects (Cisco 800 series router).
pots tone-source	Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router.

show proxy h323 calls

To list each active call on the proxy, use the **show proxy h323 calls** command in privileged EXEC mode.

show proxy h323 calls

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Examples

The following is sample output from the **show proxy h323 calls** command:

```
router# show proxy h323 calls

Call unique key = 1
  Conference ID = [277B87C0A283D111B63E00609704D8EA]
  Calling endpoint call signalling address = 55.0.0.41
  Calling endpoint aliases:
   H323_ID: ptel11@zone1.com
  Call state = Media Streaming
  Time call was initiated = 731146290 ms
```

show proxy h323 detail-call

To display the details of a particular call on a proxy, use the **show proxy h323 detail-call** command in privileged EXEC mode.

show proxy h323 detail-call *call-key*

Syntax Description

call-key	Specifies the call you want to display. The *call-key* is derived from the **show proxy h323 calls** display.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Usage Guidelines

The show proxy h323 detail-call command can be used with or without the proxy statistics enabled.

Examples

The following is sample output from the **show proxy h323 detail-call** command without the proxy statistics enabled:

```
router# show proxy h323 detail-call 1

ConferenceID = [277B87C0A283D111B63E00609704D8EA]
Calling endpoint aliases:
      H323_ID: ptel11@zone1.com
Called endpoint aliases:
      H323_ID: ptel21@zone2.com
Peer proxy call signalling address = 55.0.0.41
Time call was initiated = 731146290 ms
Inbound CRV = 144
Outbound CRV = 70
Call state = Media Streaming
H245 logical channels for call leg ptel11@zone1.com<->px1@zone.com
    Channel number = 2
        Type = VIDEO
        State = OPEN
        Bandwidth = 374 kbps
        Time created = 731146317 ms
    Channel number = 1
        Type = AUDIO
        State = OPEN
        Bandwidth = 81 kbps
        Time created = 731146316 ms
    Channel number = 2
        Type = VIDEO
```

```
            State = OPEN
            Bandwidth = 374 kbps
            Time created = 731146318 ms
        Channel number = 1
            Type = AUDIO
            State = OPEN
            Bandwidth = 81 kbps
            Time created = 731146317 ms
    H245 logical channels for call leg pte111@zone1.com<->50.0.0.41:
        Channel number = 2
            Type = VIDEO
            State = OPEN
            Bandwidth = 374 kbps
            Time created = 731146317 ms
        Channel number = 1
            Type = AUDIO
            State = OPEN
            Bandwidth = 81 kbps
            Time created = 731146316 ms
        Channel number = 2
            Type = VIDEO
            State = OPEN
            Bandwidth = 374 kbps
            Time created = 731146318 ms
        Channel number = 1
            Type = AUDIO
            State = OPEN
            Bandwidth = 81 kbps
            Time created = 731146317 ms
```

The following is sample output from the **show proxy h323 detail-call** command with the proxy statistics enabled:

```
router# show proxy h323 detail-call 1
ConferenceID = [677EB106BD0D111976200002424F832]
Calling endpoint call signalling address = 172.21.127.49
    Calling endpoint aliases:
      H323_ID: intel2
      E164_ID: 2134
Called endpoint aliases:
      H323_ID: mcs@sanjose.cisco.com
Peer proxy call signalling address = 171.68.183.199
Peer proxy aliases:
      H323_ID: proxy.sanjose.cisco.com
Time call was initiated = 730949651 ms
Inbound CRV = 2505
Outbound CRV = 67
Call state = H245 open logical channels
H245 logical channels for call leg intel2 <-> cisco7-pxy:
    Channel number = 259
      RTP stream from intel2 to cisco7-pxy
        Type = VIDEO
        State = OPEN
        Bandwidth = 225 kbps
        Time created = 730949676 ms
    Channel number = 257
      RTP stream from intel2 to cisco7-pxy
        Type = AUDIO
        State = OPEN
        Bandwidth = 18 kbps
        Time created = 730949658 ms
    Channel number = 2
      RTP stream from cisco7-pxy to intel2
        Type = VIDEO
```

```
State = OPEN
Bandwidth = 225 kbps
Time created = 730949664 ms
RTP Statistics:
  Packet Received Count = 3390
  Packet Dropped Count = 0
  Packet Out of Sequence Count = 0
  Number of initial packets used for Arrival-Spacing bin setup = 200
  min_arrival_spacing = 0(ms)  max_arrival_spacing = 856(ms)
  Average Arrival Rate = 86(ms)
  Arrival-Spacing(ms)    Packet-Count
     0                     2116
     26                    487
     52                    26
     78                    0
     104                   0
     130                   1
     156                   0
     182                   1
     208                   0
     234                   4
     260                   99
     286                   315
     312                   154
     338                   8
     364                   0
     390                   2
     416                   10
     442                   73
     468                   51
     494                   43
  ==============================
  Min Jitter = 34(ms)  Max Jitter = 408(ms)
  Average Jitter Rate = 117
  Jitter Rate(ms)    Packet-Count
     0                     0
     41                    514
     82                    2117
  Number of initial packets used for Arrival-Spacing bin setup = 200
  min_arrival_spacing = 32(ms)  max_arrival_spacing = 96(ms)
  Average Arrival Rate = 60(ms)
  Arrival-Spacing(ms)    Packet-Count
     32                    35
     34                    0
     36                    177
     38                    0
     40                    56
     42                     0
     44                    10
     46                    0
     48                    27
     50                    0
     52                    541
     54                    0
     56                    2642
     58                    1
     60                    1069
     62                    0
     64                    77
     66                    0
     68                    6
     70                    257
  ==============================
```

```
          Min Jitter = 0(ms)  Max Jitter = 28(ms)
          Average Jitter Rate = 5
          Jitter Rate(ms)   Packet-Count
             0                    1069
             3                    2720
             6                    0
             9                    804
             12                   27
             15                   10
             18                   0
             21                   56
             24                   177
             27                   35
H245 logical channels for call leg cisco7-pxy <->
proxy.sanjose.cisco.com:
    Channel number = 259
      RTP stream from cisco7-pxy to proxy.sanjose.cisco.com
        Type = VIDEO
        State = OPEN
        Bandwidth = 225 kbps
        Time created = 730949676 ms
        RTP Statistics:
          Packet Received Count = 3398
          Packet Dropped Count = 1
          Packet Out of Sequence Count = 0
          Number of initial packets used for Arrival-Spacing bin setup = 200
          min_arrival_spacing = 0(ms)  max_arrival_spacing = 872(ms)
          Average Arrival Rate = 85(ms)
          Arrival-Spacing(ms)   Packet-Count
             0                    2636
             28                   0
             56                   0
             84                   0
             112                  0
             140                  1
             168                  0
             196                  0
             224                  0
             252                  0
             280                  2
             308                  425
             336                  154
             364                  5
             392                  0
             420                  0
             448                  0
             476                  114
             504                  41
             532                  20
          ==============================
          Min Jitter = 55(ms)  Max Jitter = 447(ms)
          Average Jitter Rate = 127
          Jitter Rate(ms)   Packet-Count
             0                    0
             45                   1
             90                   2636
             135                  0
             180                  2
             225                  425
             270                  159
             315                  0
             360                  0
             405                  175
```

```
Channel number = 257
  RTP stream from cisco7-pxy to proxy.sanjose.cisco.com
    Type = AUDIO
    State = OPEN
    Bandwidth = 18 kbps
    Time created = 730949658 ms
    RTP Statistics:
      Packet Received Count = 2537
      Packet Dropped Count = 3
      Packet Out of Sequence Count = 0
      Number of initial packets used for Arrival-Spacing bin setup = 200
      min_arrival_spacing = 0(ms)  max_arrival_spacing = 32716(ms)
      Average Arrival Rate = 112(ms)
      Arrival-Spacing(ms)    Packet-Count
         0                      2191
         72                     253
         144                    31
         216                    7
         288                    3
         360                    4
         432                    4
         504                    2
         576                    1
         648                    3
         720                    2
         792                    1
         864                    2
         936                    1
         1008                   1
         1080                   1
         1152                   1
         1224                   1
         1296                   0
         1368                   28
      ==============================
      Min Jitter = 32(ms)  Max Jitter = 1256(ms)
      Average Jitter Rate = 121
      Jitter Rate(ms)    Packet-Count
         0                      284
         126                    2201
         252                    4
         378                    6
         504                    4
         630                    3
         756                    2
         882                    2
         1008                   2
         1134                   29
Channel number = 2
  RTP stream from proxy.sanjose.cisco.com to cisco7-pxy
    Type = VIDEO
    State = OPEN
    Bandwidth = 225 kbps
    Time created = 730949664 ms
Channel number = 1
  RTP stream from proxy.sanjose.cisco.com to cisco7-pxy
    Type = AUDIO
    State = OPEN
    Bandwidth = 18 kbps
    Time created = 730949661 ms
```

Related Commands

Command	Description
h323 qos	Enables QoS on the proxy.

show proxy h323 status

To display the overall status of a proxy, use the **show proxy h323 status** command in privileged EXEC mode.

show proxy h323 status

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Examples

The following is sample output from the **show proxy h323 status** command:

```
router# show proxy h323 status

  H.323 Proxy Status
       ==================
    H.323 Proxy Mode: Enabled
    Proxy interface = Serial1: UP
    Application Specific Routing: Disabled
    RAS Initialization: Complete
    Proxy aliases configured:
      H323_ID: px2
    Proxy aliases assigned by Gatekeeper:
      H323_ID: px2
    Gatekeeper multicast discovery: Disabled
    Gatekeeper:
        Gatekeeper ID: gk.zone2.com
        IP address: 70.0.0.31
    Gatekeeper registration succeeded
    T.120 Mode: BYPASS
    RTP Statistics: OFF
    Number of calls in progress: 1
```

show rawmsg

To show the raw messages owned by the required component, use the **show rawmsg** command in privileged EXEC mode.

show rawmsg {**all** | **tsp** | **vtsp** | **ccapi** | **h323**}

Syntax Description

all	All selections below.
tsp	Telephony Service Provider subsystem.
vtsp	Voice Telephony Service Provider subsystem.
ccapi	API (Application Programming Interface) used to coordinate interaction between application and call legs (telephony or IP).
h323	H.323 subsystem.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The number displayed for **show rawmsg all** should be zero, to indicate there are no memory leaks.

Examples

The following example shows how to display memory leaks from the telephony service provider:

```
show rawmsg tsp
```

Related Commands

Command	Description
isdn protocol-emulate	Configures the Layer 2 and Layer 3 port protocol of a BRI voice port or a PRI interface to emulate NT (network) or TE (user) functionality.
isdn switch type	Configures the Cisco AS5300 PRI interface to support Q.SIG signalling.
pri-group nec-fusion	Configures your NEC PBX to support FCCS.
show cdapi	Displays the CDAPI.

show settlement

To display the configuration for all settlement servers and see the specific provider and transactions, use the **show settlement** command in privileged EXEC mode.

show settlement [*provider-number*] [*transactions*]

Syntax Description

provider-number	(Optional) Displays the attributes of a specific provider.
transactions	(Optional) Displays the transaction status of a specific provider.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Examples

The following example shows information about all settlement servers configured:

```
router# show settlement
Settlement Provider 0
Type = osp
Address url = https://1.14.115.100:6556/
Encryption = all                  (default)
Max Concurrent Connections = 20   (default)
Connection Timeout = 3600 (s)     (default)
Response Timeout = 1 (s)          (default)
Retry Delay = 2 (s)               (default)
Retry Limit = 1                   (default)
Session Timeout = 86400 (s)       (default)
Customer Id = 1000
Device Id = 1000
Roaming = Disabled                (default)
Signed Token = on

Number of Connections = 0
Number of Transactions = 7
```

The following example shows transaction and state information about a specific settlement server:

```
router# show settlement 0 transactions
Transaction ID=8796304133625270342
        state=OSPC_GET_DEST_SUCCESS, index=0
        callingNumber=5710868, calledNumber=15125551212
```

Table 58 provides a description of the fields that appear with the **show settlement** command.

Table 58 show settlement Field Descriptions

Field	Description
type	Settlement provider type.
address url	URL address of the provider.
encryption	SSL encryption method.
max-connections	Maximum number of concurrent connections to provider.
connection-timeout	Connection timeout with provider (in seconds).
response-timeout	Response timeout with provider (in seconds).
retry-delay	Delay time between retries (in seconds).
retry-limit	Number of retries.
session-timeout	SSL session timeout (in seconds).
customer-id	Customer ID, assigned by provider.
device-id	Device ID, assigned by provider.
roaming	Roaming enabled.
signed-token	Indicates if the settlement token is signed by the server.

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement configuration mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

show vfc

To see the entries in the host-name-and-address cache, use the **show vfc** command in privileged EXEC mode.

show vfc *slot-number* [**technology**]

Syntax Description

slot-number	VFC slot number.
technology	(Optional) Displays the technology type of the VFC.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
10.0	This command was introduced.
12.0(2)XH	The **technology** keyword was added.

Examples

The following example shows that the card in slot 1 is a C549 DSPM:

```
5300# show vfc 1 t
Technology in VFC slot 1 is C549
```

Related Commands

Command	Description
voice-card	Configures a voice card and enters voice-card configuration mode.

show vfc cap-list

To show the current list of files on the capability list for this voice feature card (VFC), use the **show vfc cap-list** command in user EXEC mode.

show vfc *slot* **cap-list**

Syntax Description

slot	Identifies the slot where the VFC is installed. Valid entries are from 0 to 2.

Defaults

No default behavior or values.

Command Modes

User EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

To identify the specific VFC, enter the number of the slot on the chassis where the VFC resides using the *slot* argument.

Examples

The following is sample output from the **show vfc cap-list** command:

```
router> show vfc 1 cap-list

Capability List for VFC in slot 1:
1. fax-vfc-1.0.1.bin
2. bas-vfc-1.0.1.bin
3. cdc-g729-1.0.1.bin
4. cdc-g711-1.0.1.bin
5. cdc-g726-1.0.1.bin
6. cdc-g728-1.0.1.bin
7. cdc-gsmfr-1.0.1.bin
```

The first line in this output is a general description, stating that this is the capability list for the VFC residing in slot 1. Below this is a numbered list, each line of which identifies one currently installed in-service file.

Related Commands

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

show vfc default-file

To show the default files included in the default file list for this voice feature card (VFC), use the **show vfc default-file** command in user EXEC mode.

show vfc *slot* **default-file**

Syntax Description

slot	Identifies the slot where the VFC is installed. Valid entries are from 0 to 2.

Defaults

No default behavior or values.

Command Modes

User EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use the **show vfc default-file** user EXEC command to display a list of all default files for a particular voice feature card. To identify the specific VFC, enter the number of the slot on the chassis where the VFC resides using the *slot* argument.

Examples

The following is sample output from the **show vfc default-file** command:

```
router> show vfc 1 default-file

Default List for VFC in slot 1:
1. btl-vfc-1.0.13.0.bin
2. cor-vfc-1.0.1.bin
3. bas-vfc-1.0.1.bin
4. cdc-g729-1.0.1.bin
5. fax-vfc-1.0.1.bin
6. jbc-vfc-1.0.13.0.bin
```

The first line in this output is a general description, stating that this is the default list for the VFC residing in slot 1. Below this is a numbered list, each line of which identifies one default file.

Related Commands

Command	Description
show vfc cap-list	Displays the current list of files on the capability list for this VFC.
show vfc directory	Displays the list of all files residing on this VFC.
show vfc version	Displays the version of the software residing on this VFC.

show vfc directory

To show the list of all files residing on this voice feature card (VFC), use the **show vfc directory** command in user EXEC mode.

show vfc *slot* **directory**

Syntax Description

slot	Identifies the slot where the VFC is installed. Valid entries are from 0 to 2.

Defaults

No default behavior or values.

Command Modes

User EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use the **show vfc directory** user EXEC command to display a list of all of the files currently stored in Flash memory for a particular VFC. To identify the specific VFC, enter the number of the slot on the chassis where the VFC resides using the *slot* argument.

Examples

The following is sample output from the **show vfc directory** command:

```
router> show vfc 1 directory

Files in slot 1  VFC flash:
     File Name                        Size (Bytes)
1 .  vcw-vfc-mz.gsm.VCW               292628
2 .  btl-vfc-1.0.13.0.bin             4174
3 .  cor-vfc-1.0.1.bin                54560
4 .  jbc-vfc-1.0.13.0.bin             16760
5 .  fax-vfc-1.0.1.bin                64290
6 .  bas-vfc-1.0.1.bin                54452
7 .  cdc-g711-1.0.1.bin               190
8 .  cdc-g729-1.0.1.bin               21002
9 .  cdc-g726-1.0.1.bin               190
10.  cdc-g728-1.0.1.bin               22270
11.  cdc-gsmfr-1.0.1.bin              190
```

Table 59 explains the fields in the sample output.

Table 59 Show Vfc Directory Field Descriptions

Field	Description
File Name	Name of the file stored in Flash memory.
Size (Bytes)	Size of the file in bytes.

Related Commands

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

show vfc version

To show the version of the software residing on this voice feature card (VFC), use the **show vfc version** command in user EXEC mode.

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

Syntax Description

slot	Identifies the slot where the VFC is installed. Valid values are 0, 1, and 2.
dspware	Defines which DSPWare software to display.
vcware	Defines which VCWare software to display.

Defaults

No default behavior or values.

Command Modes

User EXEC

Command History

Release	Modification
11.3 NA	This command was introduced.

Usage Guidelines

Use the **show vfc version** user EXEC command to display the version of the software (either running on DSP or VFC) currently installed in Flash memory on the VFC.

Examples

The following is sample output from the **show vfc version** command:

```
router> show vfc 0 version dspware

Version of Dspware in VFC slot 0 is 0.10
```

The output from this command is a simple declarative sentence stating the version number for the selected type of software (in this example, DSPWare) for the VFC residing in the selected slot number (in this example, slot 0).

Related Commands

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

show video call summary

To display summary information about video calls and the current status of the Video Call Manager (ViCM), use the **show video call summary** command in privileged EXEC mode.

show video call summary

Syntax Description

There are no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for the Cisco MC3810.

Usage Guidelines

Use this command to quickly look at the status of current calls. In Cisco IOS Releases 12.0(5)XK and 12.0(7)T, there can be only one video call in progress.

Examples

On a Cisco MC3810, the following example displays information about the ViCM when no call is in progress on the serial interface that connects to the local video codec:

```
Router# show video call summary
Serial0:ViCM = Idle, Codec Ready
```

When a call is starting, the output looks like this:

```
Router# show video call summary
Serial0:ViCM = Call Connected
```

When a call is disconnecting, the output looks like this:

```
Router# show video call summary
Serial0:ViCM = Idle
```

Related Commands

Command	Description
show call history video record	Displays information about video calls.

show voice busyout

To display information about the voice busyout state, use the **show voice busyout** command in privileged EXEC mode.

show voice busyout

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command is only supported on the Cisco MC3810.

Examples

The following example displays the busyout information:

```
router# show voice busyout
If following network interfaces are down, voice port will be put into busyout state
ATM0
Serial0
The following voice ports are in busyout state

1/1     is forced into busyout state
1/2     is in busyout state caused by network interfaces
1/3     is in busyout state caused by ATM0
1/4     is in busyout state caused by network interfaces
1/5     is in busyout state caused by Serial0
```

Related Commands

Command	Description
busyout forced	Forces a voice port on the Cisco MC3810 multiservice concentrator into the busyout state.
busyout-monitor	Places a voice port on the Cisco MC3810 multiservice concentrator into the busyout monitor state.
busyout-seize	Changes the busyout seize procedure fro a voice port on the Cisco MC3810 multiservice concentrator.
voice-port busyout	Places all voice ports associated with a serial or ATM interface into a busyout state.

show voice call

To show the call status for all voice ports on the Cisco MC3810, use the **show voice call** command in privileged EXEC mode.

show voice call [**summary**]

Syntax Description

summary	(Optional) Specifies to show a summary of the status instead of the full detailed report.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

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

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

Examples

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

```
router# show voice call summary
1/1 (orig): eecm = ST_DIGIT_COLLECT,  LFXS= call_progress,  CPD= failure_cont

1/2 (   ): eecm = IDLE,  LFXS= idle,  CPD= idle

1/3 (   ): eecm = IDLE,  LFXS= idle,  CPD= idle

1/4 (   ): eecm = IDLE,  LFXO= idle,  CPD= idle

1/5 (   ): eecm = IDLE,  LEM= idle,  CPD= idle

1/6 (   ): eecm = IDLE,  LEM= idle,  CPD= idle
```

Table 60 explains the fields in the sample output.

Table 60 show voice call Field Descriptions

Field	Description
(orig)	Indicates the call is originating on the voice port.
eecm	Status of the End-to-End Call Manager.
LFXS	Status of the FXS line.
CPD	Status of the Call Processing Data.
LFXO	Status of the FXO line.
LEM	Status of the E&M line.

Related Commands

Command	Description
show dial-peer voice	Displays configuration information for dial peers.
show voice dsp	Displays the current status of all DSP voice channels on the Cisco MC3810 multiservice concentrator.
show voice port	Displays configuration information about a specific voice port.

show voice dsp

To show the current status of all digital signal processor (DSP) voice channels, use the **show voice dsp** command in privileged EXEC mode.

show voice dsp

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810 and Voice over IP on the Cisco 1750 router.

Examples

The following is sample output from the **show voice dsp** command for the Cisco MC3810:

```
router# show voice dsp
DSP# 0, channel# 0 G729A BUSY
DSP# 0, channel# 1 G729A BUSY
DSP# 1, channel# 2 FAX IDLE
DSP# 1, channel# 3 FAX IDLE
DSP# 2, channel# 4 NONE BAD
DSP# 2, channel# 5 NONE BAD
DSP# 3, channel# 6 NONE BAD
DSP# 3, channel# 7 NONE BAD
DSP# 4, channel# 8 NONE BAD
DSP# 4, channel# 9 NONE BAD
DSP# 5, channel# 10 NONE BAD
DSP# 5, channel# 11 NONE BAD
```

Table 61 explains the fields in the sample output.

Table 61 show voice dsp Field Descriptions

Field	Description
DSP	Number of the DSP
Channel	Number of the channel and its status.

The following is an example of the output from the command **show voice dsp** for the Cisco 1750 router:

```
router# show voice dsp
DSP#0: state IN SERVICE, 2 channels allocated
channel#0: voice port 1/0, codec G711 ulaw, state UP
channel#1: voice port 1/1, codec G711 ulaw, state UP
DSP#1: state IN SERVICE, 2 channels allocated
channel#0: voice port 2/0, codec G711 ulaw, state UP
channel#1: voice port 2/1, codec G711 ulaw, state UP
DSP#2: state RESET, 0 channels allocated
```

Table 62 explains the fields in the example output.

Table 62 show voice dsp Field Descriptions

Field	Description
DSP	Number of the DSP.
Channel	Number of the channel and its status.

Related Commands

Command	Description
show dial-peer voice	Displays configuration information for dial peers.
show voice call	Displays the call status for all voice ports on the Cisco MC3810 multiservice concentrator.
show voice port	Displays configuration information about a specific voice port.

show voice permanent-call

To display information about the permanent calls on a voice interface, use the **show voice permanent-call** command in EXEC or privileged EXEC mode.

show voice permanent-call [*voice-port*] [**summary**]

Syntax Description

voice-port	(Optional) Slot number or slot/port number of the voice interface for which you wish to display permanent call information.
summary	(Optional) Displays summary information about VoFR, VoATM, and VoHDLC ports used for permanent connections.

Defaults

No default behavior or values.

Command Modes

EXEC or Privileged EXEC

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced.

Usage Guidelines

This command is only available on the Cisco MC3810 platform.

When no parameters are specified with this command, the output displays information for all ports containing permanent calls. When a specific interface is specified, information is displayed about the permanent calls for that interface only.

Examples

The following is sample output for the **show voice permanent-call** command:

```
router# show voice permanent-call 1/1
1/1 state=connect coding=G729A payload size=30 vad=off
ec=8 (ms), cng=off fax=on digit_relay=on Seq num = off, VOFR Serial0,dlci = 550,cid = 6
TX INFO :slow-mode seq#= 25, sig pkt cnt= 19646, last-ABCD=1101
hardware-state ACTIVE signal type is CEPT/MELCAS
voice-gate CLOSED,network-path OPEN MASTER
 1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
 1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
 1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
RX INFO :slow-mode, sig pkt cnt= 19648, under-run = 0, over-run = 0
missing = 0, out of seq = 0, very late = 0
playout depth = 0 (ms), refill count = 1
 prev-seq#= 25, last-ABCD=1101, slave standby timeout 25000 (ms)
max inter-arrival time 0 (ms), current timer 384 (ms)
max timeout timer 5016 (ms), restart timeout is 0 (ms)
signaling packet fast-mode inter-arrival times (ms)
16 24 16 24 16 24 16 24 16 24 16 24 16 24 16 24
16 24 16 24 16 24 16 24 0 0 0 0 0 0 0 0
```

```
signaling playout history
1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
1101 1101 1101 1101 1101 1101 1101 1101 1101 1101
```

The following is sample output for the **show voice permanent-call summary** command:

```
router# show voice permanent-call summary
1/1 state= connect, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 880,cid = 6
1/2 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 102
1/3 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 103
1/4 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 104
1/5 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 105
1/6 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 106
1/7 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 107
1/8 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 108
1/9 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 109
1/10 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 110
1/11 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 111
1/12 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 112
1/13 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 113
1/14 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 114
1/15 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 115
1/17 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 117
1/18 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 118
1/19 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 119
1/20 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 120
1/21 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 121
1/22 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 122
1/23 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 123
1/24 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 124
1/25 state= frf11, coding=G729A, payload size=30, vad=off, ec=64, cng=off, fax=on
  digit_relay=off, VOFR Serial0:1,dlci = 990,cid = 125
```

Table 63 describes the fields shown in these displays.

Table 63 show voice permanent-call Field Descriptions

Field	Description
state	Current status of the call on this voice port.
coding	Codec type used for this call.
payload size	Size in bytes of the voice payload.
vad	Indicates whether voice activity detection is turned on or off.
ec	Echo canceller length in milliseconds.
cng	Indicates whether or not comfort noise generation is used.
fax	Indicates if fax-relay is enabled.
digit_relay	Indicates if FRF.11 Annex A DTMF digit-relay is enabled.
Seq num	Indicates whether sequence numbers are turned on or off.
VOFR	Indicates the interface used for this call.
dlci	Indicates the DLCI for this call.
cid	Indicates the DLCI subchannel for this call.
TX INFO:slow-mode	Indicates that FRF.11 Annex B packets are being sent at the slow rate defined by the signal timing keepalive period.
TX INFO:seq#	Sequence number of the last packet sent.
TX INFO:sig pkt cnt	Number of signalling packets sent by this dial peer.
TX INFO:last-ABCD	Last ABCD signalling state sent by this dial peer to the network.
hardware-state	Indicates the on-hook/off-hook state of the call when the signalling protocol in use is a supported protocol. Not valid when the signal-type is "transparent."
signal type	Indicates the type of call-control signalling used by this dial peer.
voice-gate	Indicates whether voice packets are being sent (OPEN) or not sent (CLOSED).
network-path	Indicates if any type of packet is being sent (OPEN) or not sent (CLOSED) to the network. This field will only indicate CLOSED if the port is configured as a slave using the **connection trunk answer-mode** command.
RX INFO:slow-mode	Indicates that FRF.11 Annex B packets are being received at the slow rate. Successive packets have the same sequence number.
RX INFO:sig pkt cnt	Number of slow-mode signalling packets received by this dial peer.
RX INFO:under-run	Valid for fast-mode only. Counts the number of times the signalling playout buffer became empty during FRF.11 Annex B fast-mode. In this mode, signalling packets are expected to be received every 20 milliseconds.
RX INFO:over-run	Valid for fast-mode only. Counts the number of times the signalling playout buffer became full during FRF.11 Annex B fast-mode. In this mode, signalling packets are expected to be received every 20 milliseconds.
RX INFO:missing	Indicates the number of FRF.11 Annex B packets that were counted as missing based on checking Annex B sequence numbers.

Table 63 show voice permanent-call Field Descriptions (continued)

Field	Description
RX INFO:out of seq	Indicates the number of FRF.11 Annex B packets that were counted as received in the wrong order based on checking Annex B sequence numbers.
RX INFO:very late	Indicates the number of FRF.11 Annex B packets that were received with a sequence number significantly different from the expected sequence number.
RX INFO:playout depth	Valid for fast-mode only. Shows the current FRF.11 Annex B signalling buffer playout depth in milliseconds.
RX INFO:refill count	Indicates the number of times the FRF.11 Annex B signalling playout buffer was refilled as a result of a slow-mode to fast-mode transition.
RX INFO:prev-seq#	Sequence number of the last FRF.11 Annex B signalling packet received.
RX INFO:last-ABCD	Last ABCD signalling bit pattern sent to the attached PBX (telephone network side). In the out-of-service condition, this will show the OOS pattern being sent to the PBX.
RX INFO:slave standby timeout	Value configured using the **signal timing oos standby** command for the applicable voice class permanent entry.
max inter-arrival time	Maximum interval between the arrival of fast-mode FRF.11 Annex B packets since the last time this parameter was displayed.
current timer	Time in milliseconds since the last signalling packet was received.
max timeout timer	Maximum value of the "current timer" parameter since the last time it was displayed.
restart timeout	Connection restart timeout value.
signalling packet fast-mode inter-arrival time	Shows the last several values of the fast-mode FRF.11 Annex B signalling packet inter-arrival time.
signalling playout history	Shows recent ABCD signalling bits received from the data network.

Related Commands

Command	Description
show frame-relay fragment	Displays Frame Relay fragmentation details.
show frame-relay pvc	Displays statistics about PVCs for Frame Relay interfaces.
show frame-relay vofr	Displays details about FRF.11 subchannels being used on Voice over Frame Relay DLCIs.

show voice port

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

Cisco 1750 router

show voice port *slot-number/port*

Cisco 2600 and Cisco 3600 series router

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

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**}

Cisco 7200 Series router

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

Cisco uBR924 cable access router

show voice port *number*

Syntax Description

For the Cisco 1750 router

slot-number	Slot number in the router where the VIC is installed. Valid entries are from 0 to 2, depending on the slot where it has been installed.
port	Indicates the voice port. Valid entries are 0 or 1.

For the Cisco 2600 and Cisco 3600 series router

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.
slot	The router location where the voice port adapter is installed. Valid entries are from 0 to 3.

port	Indicates the voice interface card location. Valid entries are 0 or 3.
dso-group-no	Indicates the defines DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

For the Cisco MC3810:

slot/port	(Optional) Displays information for only the voice port you specify with the *slot/port* designation. The *slot* argument specifies the slot number in the Cisco router where the voice interface card is installed. The only valid entry is 1. The *port* argument 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* argument is 0 to 9999. Valid entries for the *slot* argument is 0 to 11. Valid entries for the *port* argument is 0 to 11.
shelf/slot/parent:port	Specifies the T1 controller on the T3 card. Valid entries for the *shelf* argument is 0 to 9999. Valid entries for the *slot* argument is 0 to 11. Valid entries for the *port* argument is 1 to 28. The value for the *parent* argument is always 0.
:D	Indicates the D channel associated with ISDN PRI.

For the Cisco 7200 series router

slot	The router location where the voice port adapter is installed. Valid entries are from 0 to 3.
port	Indicates the voice interface card location. Valid entries are 0 or 1.
dso-group-no	Indicates the defines DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.
slot-number	Indicates the 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	Indicates the subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1.
port	Indicates the voice port number. Valid entries are 0 or 1.

For the Cisco uBR924 cable access router

number	Indicates the RJ-11 connectors installed in the Cisco uBR924. Valid entries are 0 (which corresponds to the RJ-11 connector labeled V1) and 1 (which corresponds to the RJ-11 connector labeled V2.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3 MA and 12.0(3)T	Port-specific values for the Cisco MC3810 were added.
12.0(5)XE and 12.0(7)T	Additional syntax was created for digital voice to allow specification of the DS0 group. This command applies to VoIP on the Cisco 7200 series.
12.0(5)XK and 12.0(7)T	Additional syntax was created for digital voice on the Cisco 2600 and Cisco 3600 series to allow specification of the DS0 group.
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.

The **ds0-group** command automatically creates a logical voice port that is numbered as follows on Cisco 7200 series routers and the Cisco 2600 and Cisco 3600 series routers: *slot/port:ds0-group-no*. Although only one voice port is created for each group, applicable calls are routed to any channel in the group.

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 example displays voice port configuration information for the digital voice port 0 located in slot 1, DS0 group 1 on the Cisco 3600 series:

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

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

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 64 explains the fields in the sample output.

Table 64 show voice port Field Descriptions

Field	Description
Administrative State	Administrative state of the voice port.
Alias	User-supplied alias for this voice port.
Analog interface A-D gain offset	Offset of the gain for analog-to-digital conversion.
Analog interface D-A gain offset	Offset of the gain for digital-to-analog conversion.
Clear Wait Duration Timing	Time of inactive seizure signal to declare call cleared.
Coder Type	Voice compression mode used.
Companding Type	Companding standard used to convert between analog and digital signals in PCM systems.
Connection Mode	Connection mode of the interface.
Connection Number	Full E.164 telephone number used to establish a connection with the trunk or PLAR mode.
Currently Processing	Type of call currently being processed: none, voice, or fax.
Delay Duration Timing	Maximum delay signal duration for delay dial signalling.
Delay Start Timing	Timing of generation of delayed start signal from detection of incoming seizure.

Table 64 show voice port Field Descriptions (continued)

Field	Description
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 nonlinear processing is enabled for this port.
Number of signalling protocol errors	Number of signalling protocol errors.
Operations State	Operation state of the port.
Operation Type	Operation of the E&M signal: 2-wire or 4-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.

Table 64 show voice port Field Descriptions (continued)

Field	Description
Ring Cadence	Configured ring cadence for this interface.
Ring Frequency	Configured ring frequency for this interface.
Ring Ground Status	Ring ground indication.
Ringing Time Out	Ringing time out duration.
Signal Type	Type of signalling 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 signalling.
Wink Wait Duration Timing	Maximum wink wait duration for wink start signalling.

The following example displays voice port configuration information for the digital voice port 0 located in slot 1, DS0 group 1:

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

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

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 65 explains the fields in the sample output.

Table 65 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.
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 nonlinear 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.

Table 65 show voice port Field Descriptions for the Cisco AS5800 (continued)

Field	Description
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 contents of the active call table.
show call history voice	Displays the contents of the call history table.
show dial-peer voice	Displays configuration information and call statistics 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 voice feature card (VFC) or all VFCs, use the **show vrm active_calls** command in privileged EXEC mode.

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 66 explains the fields in the sample output.

Table 66 show vrm active_calls 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 time slot	Time division multiplexing time slot.
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 66 show vrm active_calls 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 digital signal processor (DSP) or a brief summary display for all voice feature cards (VFCs), use the **show vrm vdevices** command in privileged EXEC mode.

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 vdevices** command 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 sending 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 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 vdevices** 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 67 explains the fields in the sample output.

Table 67 show vrm vdevices 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.

Table 67 show vrm vdevices Field Descriptions (continued)

Field	Description
TDM time slot	Time division multiplexing time slot.
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.
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 68 explains the fields in the sample output.

Table 68 show vrm vdevices 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. Possible field values are: • 0 = VFC is in classic mode • 1 = VFC is in plus mode. This field should always be 1.
num_of_tdm_time slots	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.
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.

Table 68 show vrm vdevices summary Field Descriptions (continued)

Field	Description
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
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

Table 68 show vrm vdevices summary Field Descriptions (continued)

Field	Description
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.

shut

To shut down a set of digital signal processors (DSPs) on the Cisco 7200 series router, use the **shut** command in DSP configuration mode. Use the **no** form of this command to put DSPs back in service.

shut *number*

no shut *number*

Syntax Description

number	Indicates the number of DSPs to be shutdown.

Defaults

No shut

Command Modes

DSP configuration

Command History

Release	Modification
12.0(5)XE	This command was introduced.

Usage Guidelines

This command applies to Voice over IP on the Cisco 7200 series routers.

Examples

The following example shuts down two sets of DSPs:

```
router(config-dspfarm)# shut 2
```

shutdown (dial-peer configuration)

To change the administrative state of the selected dial peer from up to down, use the **shutdown** command in dial-peer configuration mode. Use the **no** form of this command to change the administrative state of this dial peer from down to up.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Defaults

no shutdown

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

When a dial peer is shut down, you cannot initiate calls to that peer.

Examples

The following example changes the administrative state of voice telephony (POTS) dial peer 10 to down:

```
configure terminal
 dial-peer voice 10 pots
 shutdown
```

shutdown (DS1 link)

To shut down a DS1 link (send a Blue Alarm), use the **shutdown** command in controller configuration mode. Use the **no** form of the command to activate the DS1 (cancel the sending of the Blue Alarm).

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Defaults

no shutdown

Command Modes

Controller configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Examples

The following example shuts down a DS1 link on controller T1 0:

```
controller T1 0
 shutdown
```

shutdown (MCM)

To disable the gatekeeper, use the **shutdown** gatekeeper configuration command. To enable the gatekeeper, use the **no** form of this command.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled (shut down)

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Usage Guidelines

The gatekeeper does not have to be enabled before you can use the other gatekeeper configuration commands. In fact, it is recommended that you complete the gatekeeper configuration before bringing up the gatekeeper because some characteristics may be difficult to alter while the gatekeeper is running, as there may be active registrations or calls.

While the no shutdown command enables the gatekeeper, it does not make it operational. The two exceptions to this are:

- If no local zones are configured, a **no shutdown** command places the gatekeeper in INACTIVE mode waiting for a local zone definition.
- If local zones are defined to use an HSRP virtual address, and the HSRP interface is in STANDBY mode, the gatekeeper goes into HSRP STANDBY mode. Only when the HSRP interface is ACTIVE will the gatekeeper go into the operational UP mode.

Examples

The following command disables a gatekeeper:

```
shutdown
```

shutdown (settlement)

To activate a settlement provider, use the **no shutdown** command in settlement configuration mode. Use the **shutdown** command to deactivate the settlement provider.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Defaults

The default status of a settlement provider is deactivated. The settlement provider is down.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

Use the **no shutdown** command at the end of the configuration of a settlement server to bring up the provider. This command activates the provider. Otherwise, transactions will not go through the provider to be audited and charged. Use **shutdown** to deactivate the provider.

Examples

The following example enables a settlement server:

```
settlement 0
 no shutdown
```

The following example disables a settlement server:

```
settlement 0
 shutdown
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-delay	Sets the time between attempts to connect with the settlement provider.

Command	Description
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement configuration mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

shutdown (voice-port configuration)

To take the voice ports for a specific voice interface card offline, use the **shutdown** command in voice-port configuration mode. Use the **no** form of this command to put the ports back in service.

shutdown

no shutdown

Syntax Description

This command has no arguments or keywords.

Defaults

shutdown

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

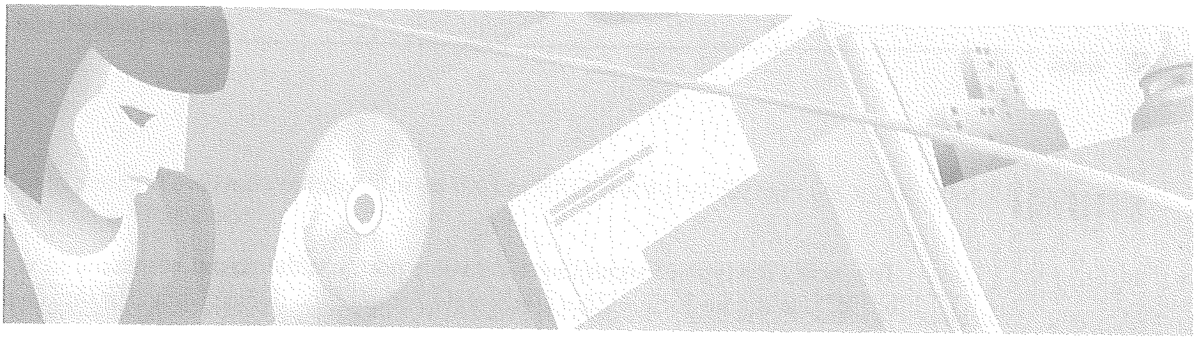
When you enter the **shutdown** command, all ports on the voice interface card are disabled. When you enter the **no shutdown** command, all ports on the voice interface card are enabled. A telephone connected to an interface will hear dead silence when a port is shut down.

Examples

The following example takes voice port 1/1/0 on the Cisco 3600 series offline:

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

Note The preceding configuration example shuts down both voice ports 1/1/0 and 1/1/1.



Multiservice Applications Commands: Si through Z

This book documents commands used to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features. Commands in this book are listed alphabetically. For information on how to configure Voice over ATM, Voice over Frame Relay, Voice over HDLC, Voice over IP, video, head-end universal broadband features, and subscriber-end universal broadband features, refer to the *Cisco IOS Multiservice Applications Configuration Guide*.

signal

To specify the type of signalling for a voice port, use the **signal** command in voice-port configuration mode. Use the **no** form of this command to restore the default value for this command.

For FXO and FXS voice ports:

signal {**loop-start** | **ground-start**}

no signal {**loop-start** | **ground-start**}

For E&M voice ports:

signal {**wink-start** | **immediate** | **delay-dial**}

no signal {**wink-start** | **immediate** | **delay-dial**}

Syntax Description

loop-start	Specifies loop start signalling. Used for FXO and FXS interfaces. With loop start signalling only one side of a connection can hang up. This is the default setting for FXO and FXS voice ports.
ground-start	Specifies ground start signalling. Used for FXO and FXS interfaces. Ground Start allows both sides of a connection to place a call and to hang up.
wink-start	Indicates that the calling side seizes the line by going off-hook on its E-lead then waits for a short off-hook "wink" indication on its M lead from the called side before sending address information as DTMF digits. Used for E&M tie trunk interfaces. This is the default setting for E&M voice ports.
immediate	Indicates that the calling side seizes the line by going off-hook on its E-lead and sends address information as DTMF digits. Used for E&M tie trunk interfaces.
delay-dial	Indicates that the calling side seizes the line by going off-hook on its E-lead. After a timing interval, the calling side looks at the supervision from the called side. If the supervision is on-hook, the calling side starts sending information as DTMF digits; otherwise, the calling side waits until the called side goes on-hook and then starts sending address information. Used for E&M tie trunk interfaces.

Defaults

loop-start for FXO and FXS interfaces; wink-start for E&M interfaces

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to analog voice ports only.

Configuring the **signal** command for an FXS or FXO voice port changes the signal value for both voice ports on a VPM card.

Note If you change the signal type for an FXO voice port on Cisco 3600 series routers, you need to move the appropriate jumper in the voice interface card of the voice network module. For more information about the physical characteristics of the voice network module, refer to the installation documentation, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your voice network module.

Configuring this command for an E&M voice port changes only the signal value for the selected voice port. In either case, the voice port must be shut down and then activated before the configured values will take effect.

Some PBXs will miss initial digits if the E&M voice port is configured for Immediate signalling. If this occurs, use Delay-Dial signalling instead. Some non-Cisco devices have a limited number of DTMF receivers. This type of equipment must delay the calling side until a DTMF receiver is available.

Examples

The following example configures ground start signalling on the Cisco 3600 series, which means that both sides of a connection can place a call and hang up, as the signalling type for a voice port:

```
configure terminal
 voice-port 1/1/1
 signal ground-start
```

signal keepalive

To configure the keepalive signalling packet interval for Cisco trunks and FRF.11 trunks, use the **signal keepalive** command in voice-class configuration mode. Use the **no** form of this command to restore the default value.

signal keepalive *number*

no signal keepalive *number*

Syntax Description

number	Specifies the keepalive signalling packet interval in seconds. The valid range is from 1 to 65535 seconds.

Defaults

5 seconds

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

Before configuring the keepalive signalling interval, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. The voice class must then be assigned to a dial peer.

Examples

The following example, beginning in global configuration mode, sets the keepalive signalling interval to 3 seconds for voice class 10.

```
voice class permanent 10
 signal keepalive 3
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of the call.
signal timing oos	Configures the signal timing parameter for the OOS state of the call.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice class permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal pattern

To configure the ABCD bit pattern for Cisco trunks and FRF.11 trunks, use the **signal pattern** command in voice-class configuration mode. Use the **no** form of this command to remove the signal pattern setting from the voice class.

signal pattern {**idle receive** | **idle transmit** | **oos receive** | **oos transmit**} *word*

no signal pattern {**idle receive** | **idle transmit** | **oos receive** | **oos transmit**} *word*

Syntax Description

idle receive	Specifies that the signal pattern applies to the idle state of the call for receive bits. The receive direction is from the network to the PBX.
idle transmit	Specifies that the signal pattern applies to the idle state of the call for transmit bits. The transmit direction is from the PBX to the network.
oos receive	Specifies that the signal pattern applies to the out-of-service state of the call for receive bits.
oos transmit	Specifies that the signal pattern applies to the out-of-service state of the call for transmit bits.
word	The ABCD bit pattern. Valid values are from 0000 to 1111.

Defaults

No signal pattern is defined.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

Before configuring the signalling pattern, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. The voice class must then be assigned to a dial peer.

This command must be entered twice. When you enter the command to specify the signalling pattern for the idle transmit state, you must reenter the command to specify the signalling pattern for the idle receive state.

The idle state of a call is normally based on both the transmit and receive idle patterns matching the signalling state in the signalling packets. If only one direction is configured (transmit or receive), the idle state will be detected based only on the direction that is configured. The out-of-service (OOS) transmit pattern is matched against the signalling state from the PBX (and transmitted to the network). This is used in conjunction with either the suppress-voice timing parameter or the suppress-all parameter.

The OOS receive pattern is the pattern *sent* to the PBX if the signal timing oos timeout timer expires during which no signalling packets are received from the network. The OOS receive pattern is not used for pattern matching against the signalling packets received from the network. The receive packets directly indicate an OOS condition by setting the AIS alarm indication bit in the packet.

To "busy out" a PBX if the network connection fails, set the OOS receive pattern to match the seized state (busy), then set the **signal timing oos** timeout value. When the timeout value expires and no signalling packets have been received, the router will send the OOS receive pattern to the PBX.

Use the busy seized pattern only if the PBX does not have a special pattern specifically intended to indicate an OOS state. If the PBX does have a specific OOS pattern, use that pattern instead.

Examples

The following example, beginning in global configuration mode, configures the signalling bit pattern for the idle receive and transmit states:

```
voice class permanent 10
 signal keepalive 3
 signal pattern idle receive 0101
 signal pattern idle transmit 0101
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

The following example, beginning in global configuration mode, configures the signalling bit pattern for the out-of-service receive and transmit states:

```
voice class permanent 10
 signal keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
signal keepalive	Configures the keepalive signalling packet interval for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of the call.
signal timing oos	Configures the signal timing parameter for the OOS state of the call.
signal-type	Sets the signalling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice class permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing idle suppress-voice

To configure the signal timing parameter for the idle state of the call, use the **signal timing idle suppress-voice** command in voice-class configuration mode. Use the **no** form of this command to restore the default value.

signal timing idle suppress-voice *seconds*

no signal timing idle suppress-voice *seconds*

Syntax Description

seconds	Duration of the idle state in seconds before the voice traffic is stopped. The valid range is from 0 to 65535.

Defaults

No signal timing idle suppress-voice timer is configured.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

Before configuring the signal timing idle suppress-voice timer, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. The voice class must then be assigned to a dial peer.

This command is used when the **signal-type** command is set to **transparent** in the dial peer for the Cisco trunk or FRF.11 trunk connection. The Cisco MC3810 stops sending voice packets when the timer expires. Signalling packets are still sent.

Examples

The following example, beginning in global configuration mode, sets the signal timing idle suppress-voice timer to 5 for the idle state on voice class "10."

```
voice class permanent 10
 signal keepalive 3
 signal pattern idle receive 0101
 signal pattern idle transmit 0101
 signal timing idle suppress-voice 5
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
 signal-type transparent
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
signal keepalive	Configures the keepalive signalling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing oos	Configures the signal timing parameter for the OOS state of the call.
signal-type	Sets the signalling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice class permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing oos

To configure the signal timing parameter for the out-of-service (OOS) state of the call, use the **signal timing oos** command in voice-class configuration mode. Use the **no** form of this command to restore the default value.

signal timing oos {**restart** | **slave-standby** | **suppress-all** | **suppress-voice** | **timeout**} *seconds*

no signal timing oos {**restart** | **slave-standby** | **suppress-all** | **suppress-voice** | **timeout**} *seconds*

Syntax Description

restart	If no signalling packets are received for this period, the permanent voice connection will be torn down and an attempt to achieve reconnection will be made.
slave-standby	If no signalling packets are received for this period, a slave port returns to its initial standby state. This option applies only to slave ports (ports configured using the **connection trunk** *number* **answer-mode** command).
suppress-all	If the transmit OOS pattern (from the PBX to the network) matches for this period of time, the router stops sending all packets to the network.
suppress-voice	If the transmit OOS pattern (from the PBX to the network) matches for this period of time, the router stops sending voice packets to the network. Signalling packets continue to be sent with the alarm indication set (AIS).
timeout	If no signalling packets are received for this period of time, the router sends the configured receive OOS pattern to the PBX. Also, the router stops sending voice packets to the network. Use this option to perform busyout to the PBX.
seconds	Duration in seconds for the above settings. The valid range is from 0 to 65535.

Defaults

No signal timing OOS pattern parameters are configured.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

Before configuring signal timing OOS parameters, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. The voice class must then be assigned to a dial peer.

You can enter several values for this command. However, the **suppress-all** and **suppress-voice** options are mutually exclusive.

Examples

The following example, beginning in global configuration mode, configures the signal timeout parameter for the out-of-service state on voice class 10. The **signal timing oos timeout** command is set to 60 seconds.

```
voice-class permanent 10
 signal-keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 signal timing oos timeout 60
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
signal keepalive	Configures the keepalive signalling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of the call.
signal-type	Sets the signalling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.

signal-type

To set the signalling type to be used when connecting to a dial peer, use the **signal-type** dial-peer configuration command. To return to the default signal-type, use the **no** form of this command.

Cisco 2600 series and 3600 series routers

signal-type {**cas** | **ext-signal**}

no signal-type {**cas** | **ext-signal**}

Cisco MC3810

signal-type {**cas** | **cept** | **ext-signal** | **transparent**}

no signal-type {**cas** | **cept** | **ext-signal** | **transparent**}

Syntax Description

cas	North American EIA-464 channel-associated signalling (robbed-bit signalling).
cept	Provides a basic E1 ABCD signalling protocol. Used primarily for E&M interfaces. When used with FXS/FXO interfaces, this protocol is equivalent to MELCAS.
ext-signal	External signalling. The digital signal processor (DSP) does not generate any signalling frames. Use this option when there is an external signalling channel (for example, CCS) or when you need to have a permanent "dumb" voice pipe.
transparent	Selecting this option produces different results depending on whether you are using a digital voice module (DVM) or an analog voice module (AVM). For a DVM: The ABCD signalling bits are copied from or transported through the T1/E1 interface "transparently," without modification or interpretation. This enables the MC3810 to handle arbitrary or unknown signalling protocols. For an AVM: It is not possible to provide "transparent" behavior because the Cisco MC3810 must interpret the signalling information in order to read/write the correct state to the analog hardware. This option is mapped to be equal to "cas."

Defaults

cas

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.
12.0(4)T	Support was added for the Cisco 7200 series routers.

Usage Guidelines

This command applies to VoFR, VoATM, and VoHDLC dial peers. It is used with permanent connections only (Cisco trunks and FRF.11 trunks), not with switched calls.

This command is used to inform the local telephony interface of the type of signalling it should expect to receive from the far-end dial peer. To turn signalling off at this dial peer, select the **ext-signal** option. If signalling is turned off and there are no external signalling channels, a "hot" line exists, enabling this dial peer to connect to anything at the far end.

On the Cisco 2600 series and 3600 series routers, there are only two possible settings for trunks (VoFR dial peers only):

- Signalling is enabled (the default, North American EIA-464 CAS signalling)
- Signalling is disabled (signal-type ext-signal)

When you connect an FXS to another FXS, or if you have anything other than an FXS/FXO or E&M/E&M pair, the appropriate signalling type on Cisco 2600 series and 3600 series routers is **ext-signal** (disabled).

On the Cisco MC3810, there are two additional signal-type settings:

- Signal-type cept (European)
- Signal-type transparent (pass-through)

If you have a digital E1 connection at the remote end that is running cept/MELCAS signalling and you then trunk that across to an analog port, you should make sure that you configure both ends for the **cept** signal-type.

If you have a T1 or E1 connection at both ends and the T1/E1 is running a signalling protocol that is neither EIA-464nor cept/MELCAS, you might want to configure the signal-type for the transparent option in order to pass through the signalling.

Examples

The following example shows how to disable signalling on a Cisco 2600 series or 3600 series router or on a Cisco MC3810 multiservice access concentrator for VoFR dial peer 200, starting from global configuration mode:

```
dial-peer voice 200 vofr
 signal-type ext-signal
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a Voice over Frame Relay dial peer.
connection	Specifies a connection mode for a voice port.
destination-pattern	Specifies either the prefix, the full E.164 telephone number, or an ISDN directory number (depending on the dial plan) to be used for a dial peer.
dtmf-relay	Enables the generation of FRF.11 Annex A frames for a dial peer.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.
session protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session target	Specifies a network-specific address for a specified dial peer or destination gatekeeper.
sequence-numbers	Enables the generation of sequence numbers in each frame generated by the digital signal processor (DSP) for Voice over Frame Relay applications.

snmp enable peer-trap poor-qov

To generate poor quality of voice notification for applicable calls associated with VoIP dial peers, use the **snmp enable peer-trap poor-qov** command in dial-peer configuration mode. Use the **no** form of this command to disable this notification.

snmp enable peer-trap poor-qov

no snmp enable peer-trap poor-qov

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **snmp enable peer-trap poor qov** command to generate poor quality of voice notifications for applicable calls associated with this dial peer. If you have an SNMP manager that will use SNMP messages when voice quality drops, you might want to enable this command. Otherwise, you should disable this command to reduce unnecessary network traffic.

Examples

The following example enables poor quality of voice notifications for calls associated with VoIP dial peer 10:

```
dial-peer voice 10 voip
 snmp enable peer-trap poor-qov
```

Related Commands

Command	Description
snmp-server enable traps	Enables a router to send SNMP traps and informs.
snmp trap link-status	Enables SNMP trap messages to be generated when a specific port is brought up or down.

supervisory disconnect

To enable a supervisory disconnect signal on FXO ports, use the **supervisory disconnect** command in voice-port configuration mode. Use the **no** form of this command to disable the supervisory disconnect signal.

supervisory disconnect

no supervisory disconnect

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command indicates whether or not supervisory disconnect signalling is available on the FXO port. Supervisory disconnect signalling is a power denial from the switch lasting at least 350 milliseconds. When this condition is detected, the system interprets this as a disconnect indication from the switch and clears the call.

You should configure no supervisory disconnect on the voice port if there is no supervisory disconnect available from the switch.

Note If there is no disconnect supervision on the voice port, the interface could be left active if the caller abandons the call before the far end answers. After the router collects the dialed digits but before the called party answers, the router starts a tone detector. Within this time window, the tone detector listens for signals (such as a fast busy signal) that occur if the originating caller hangs up. If this occurs, the router will interpret those tones as a disconnect indication and close the window.

Examples

The following example configures supervisory disconnect on a Cisco 3600 series voice port:

```
voice-port 2/1/0
 supervisory disconnect
```

The following example configures supervisory disconnect on a Cisco MC3810 voice-port:

```
voice-port 1/1
 supervisory disconnect
```

tdm-group

To configure a list of time slots for creating clear channel groups (pass-through) for time-division multiplexing (TDM) cross-connect, use the **tdm-group** command in controller configuration mode. Use the **no** form of this command to delete a clear channel group.

tdm-group *tdm-group-no timeslot timeslot-list* [**type** {**e&m** | **fxs** [**loop-start** | **ground-start**] | **fxo** [**loop-start** | **ground-start**] | **fxs-melcas** | **fxo-melcas** | **e&m-melcas**}]

no tdm-group *tdm-group-no timeslot timeslot-list* [**type** {**e&m** | **fxs** [**loop-start** | **ground-start**] | **fxo** [**loop-start** | **ground-start**] | **fxs-melcas** | **fxo-melcas** | **e&m-melcas**}]

Syntax Description

tdm-group-no	Time Division Multiplexing (TDM) group number.
timeslot	Timeslot number.
timeslot-list	Timeslot list. The valid range is from 1-24 for T1, and from 1-15 and 17-31 for E1.
type	(Optional) (Valid only when the **mode cas** command is enabled.) Specifies the voice signalling type of the voice port. If configuring a TDM group for data traffic only, do not specify the **type** keyword. Choose from one of the following options: • **e&m**—for E&M signalling • **fxo**—for Foreign Exchange Office signalling (optionally, you can also specify loop-start or ground-start) • **fxs**—for Foreign Exchange Station signalling (optionally, you can also specify loop-start or ground-start) • **e&m-melcas**—for E&M Mercury Exchange Limited Channel-Associated Signalling (MEL CAS) • **fxs-melcas**— for Foreign Exchange Station MEL CAS • **fxo-melcas**—for Foreign Exchange Office MEL CAS The **melcas** options apply only to E1 lines and are used primarily in the United Kingdom.

Defaults

No TDM group is configured.

Command Modes

Controller configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Note Channel groups, CAS voice groups, and TDM groups all use group numbers. All group numbers configured for channel groups, CAS voice groups and TDM groups must be unique on the local Cisco MC3810 concentrator. For example, you cannot use the same group number for a channel group and for a TDM group.

Examples

The following example configures TDM group number 20 on controller T1 1 to support FXO ground-start:

```
controller T1 1
 mode cas
 tdm-group 20 20 type fxs ground-start
```

Related Commands

Command	Description
mode (Voice over ATM)	Sets the mode of the T1/E1 controller and enters specific configuration commands for each mode type in Voice over ATM.

tech-prefix

To specify a particular technology prefix be prepended to the destination pattern of a specific dial peer, use the **tech-prefix** command in dial-peer configuration mode. Use the **no** form of this command to disable the defined technology prefix for this dial peer.

tech-prefix *number*

no tech-prefix *number*

Syntax Description

number	Defines the numbers used as the technology prefix. Each technology prefix can contain up to 11 characters. Although not strictly necessary, a pound (#) symbol is frequently used as the last digit in a technology prefix. Valid characters are 0 though 9, the pound (#) symbol, and the asterisk (*).

Defaults

No technology prefix is defined.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced.

Usage Guidelines

Technology prefixes are used to distinguish between gateways having specific capabilities within a given zone. In the exchange between the gateway and the gatekeeper, the technology prefix is used to select a gateway after the zone has been selected. Use the **tech-prefix** command to define technology prefixes.

Technology prefixes can be used as a discriminator so that the gateway can tell the gatekeeper that a certain technology is associated with a particular call (for example, 15# could mean a fax transmission), or it can be used like an area code for more generic routing. No standard defines what the numbers in a technology prefix mean; by convention, technology prefixes are designated by a pound (#) symbol as the last character.

In most cases, there is a dynamic protocol exchange between the gateway and the gatekeeper that enables the gateway to inform the gatekeeper about technology prefixes and where to forward calls. If, for some reason, that dynamic registry feature is not in effect, you can statically configure the gatekeeper to query the gateway for this information by configuring the **gw-type-prefix** command on the gatekeeper. Use the **show gatekeeper gw-type-prefix** to display how the gatekeeper has mapped the technology prefixes to local gateways.

Note Cisco gatekeepers use the asterisk (*) as a reserved character. If you are using Cisco gatekeepers, do not use the asterisk as part of the technology prefix.

Examples

The following example defines a technology prefix of 14# for the specified dial peer. In this example, the technology prefix means that the H.323 gateway will ask the RAS gatekeeper to direct calls using the technology prefix of 14#.

```
dial-peer voice 10 voip
 destination-pattern 14...
 tech-prefix 14#
```

Related Commands

Command	Description
gw-type-prefix	Configures a technology prefix in the gatekeeper.
show gatekeeper gw-type-prefix	Displays the gateway technology prefix table.

test cable awacs

To test a cable modem card in a Cisco uBR7200 series cable router using onboard spectrum management hardware, use the **test cable awacs** command in EXEC mode.

test cable *slot/port* **awacs**

Syntax Description

slot/port	Specifies the slot and port number of the cable modem card for which information is to be collected.

Defaults

No default behavior or values.

Command Modes

EXEC

Command History

Release	Modification
12.0(4)XI	This command was introduced.

Usage Guidelines

This command is supported only on the MC16S cable modem card.

Examples

The following example tests a cable modem card:

```
test cable 6/3 awacs
```

test cable atp cable

To run the acceptance test procedure on a Cisco uBR7200 series cable modem card, use the **test cable atp cable** command in privileged EXEC mode.

test cable atp cable *slot/port MAC-address category test-id*

Syntax Description

slot/port	Specifies the upstream cable interface by slot and port number.
MAC-address	Specifies the MAC address of the cable modem.
category	Specifies the test category as being *mac* for MAC tests or *mp* for MAC-PHY tests. Valid MAC tests are 1 through 15. Valid MAC-PHY tests are 4 through 7. These categories of tests are described in the ATP documentation.
test-id	Identifies a test specified in the automatic test procedure (ATP) documentation. The ATP documentation describes the collection of tests and the categories into which these tests are divided.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(6) NA	This command was introduced.

Usage Guidelines

You should read and understand the ATP documentation before using this command.

The ATP tests are organized into categories such as PHY, MP, MAC, and so forth. Tests within each category are labeled MP01, MP02, ..., MAC01, MAC02, and so forth. If you run a test from the CLI, you can omit the leading zero in the test ID.

In this release, Cisco supports only a subset of all of the tests.

Examples

The following example tests the upstream cable interface located in slot 2/port 0 at MAC address 1.1.1. The test specified is MAC-PHY test 4 (MP-04).

```
CMTS01# test cable atp cable 2/0 1.1.1 mp 4
Running Upstream Channel Change (MP-04)
Testing MP_04_UCD_FREQ_CHANGE
Setting the upstream to 30MHz through UCD.
Waiting 30 seconds for new frequency to be effective.
05:18:46: %UBR7200-5-USFREQCHG: Interface Cable2/0 Port U-1, frequency changed to 30.000
MHz
Conducting connectivity test.
```

Some tests, such as the one shown below, produce voluminous output:

```
CMTS01# test cable atp c6/0 0010.7b43.aab9 8
Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 19.1.25.195, timeout is 2 seconds:
.!!!!
Success rate is 80 percent (4/5), round-trip min/avg/max = 8/8/8 ms
*** 1-1. Normal TLV order UCD test started.
Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 19.1.25.195, timeout is 2 seconds:
!!!!!
Success rate is 100 percent (5/5), round-trip min/avg/max = 8/26/100 ms
*** 1-1. Normal TLV order UCD test passed.
Continue to next step?[confirm]
*** 1-2. Reversed TLV order UCD test started.
Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 19.1.25.195, timeout is 2 seconds:
!!!!!
...
!!!!!
Success rate is 100 percent (5/5), round-trip min/avg/max = 8/27/104 ms
*** 2-6. negative burst descriptor(type 129, len 1; in Request msg) test passed.

Continue to next step?[confirm]
*** 2-7. undefined burst descriptor(type 12; in Short Data msg) test started.
Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 19.1.25.195, timeout is 2 seconds:
!!!!!
Success rate is 100 percent (5/5), round-trip min/avg/max = 8/26/100 ms
*** 2-7. undefined burst descriptor(type 12; in Short Data msg) test passed.
Continue to next step?[confirm]
*** 2-8. Null burst descriptor(len 0, type 12; in Short Data msg) test started.
Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 19.1.25.195, timeout is 2 seconds:
!!!!!
Success rate is 100 percent (5/5), round-trip min/avg/max = 8/27/104 ms
*** 2-8. Null burst descriptor(len 0, type 12; in Short Data msg) test passed.
Continue to next step?[confirm]
*** 2-9. negative burst descriptor(type 129, len 1; in Short Data msg) test
started.
Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 19.1.25.195, timeout is 2 seconds:
!!!!!
Success rate is 100 percent (5/5), round-trip min/avg/max = 8/26/100 ms
*** 2-9. negative burst descriptor(type 129, len 1; in Short Data msg) test
passed.
Continue to next step?[confirm]
*** 3-1. Number of burst profiles test(#burst desc.in UCD > # burst profiles in
MAP) started.
Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 19.1.25.195, timeout is 2 seconds:
!!!!!
...
!!!!!
Success rate is 100 percent (5/5), round-trip min/avg/max = 8/26/100 ms
*** 4-1. Long Grant without max. burst size test(Short Grant size=1) passed.
Continue to next step?[confirm]
*** 5-6. UCD count less than MAP change count test started.
 UCD count:19, next MAP change count:20
 Station maintenance req failed
 UCD count:19, next MAP change count restored:19
*** 5-6. UCD count less than MAP change count test passed.
Continue to next step?[confirm]
*** 5-7. Stopping UCD test started.
```

```
CM T1 timeout and reset (y/n)?[confirm]
 wait for CM to come up again.
Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 19.1.25.195, timeout is 2 seconds:
.!!!!
Success rate is 80 percent (4/5), round-trip min/avg/max = 8/34/112 ms
*** 5-7. Stopping UCD test passed.
Continue to next step?[confirm]
...
```

test vrm busyout

To busy out a specific digital signal processor (DSP) or channels on a specific DSP, use the **test vrm busyout** command in privileged EXEC mode.

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

Syntax Description

slot-number	Number identifying 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 digital signal processor (DSP) or a range of DSPs on a specific VFC. In addition, you can use this command 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:

```
test vrm busyout 4 all
```

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

```
test vrm busyout 4 1 3
```

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

```
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 digital signal processor (DSP), use the **test vrm reset** command in privileged EXEC mode.

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

Syntax Description

slot-number	Number identifying 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 terminate active calls [confirm)
 Reset command sent to voice card 4 for voice device 2.
```

test vrm unbusyout

To restore activity to a busied-out digital signal processor (DSP) or busied-out channels on a digital signal processor (DSP), use the **test vrm unbusyout** command in privileged EXEC mode.

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

Syntax Description

slot-number	Number identifying 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 command 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:

```
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:

```
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:

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

Related Commands

Command	Description
test vrm busyout	Busyouts a specific DSP or channels on a specific DSP.

timeouts call-disconnect

To configure the call disconnect time-out value for a specified voice port, use the **timeouts call-disconnect** command in voice-port configuration mode. Use the **no** form of this command to restore the default value.

timeouts call-disconnect *seconds*

no timeouts call-disconnect

Syntax Description

seconds	Sets the call-disconnect time-out duration in seconds. Valid values are from 0 to 120.

Defaults

60 seconds.

Command Modes

V.oice-port configuration

Command History

Release	Modification
11.3(9)T and 12.0(4)T	This command was introduced.

Usage Guidelines

This command applies to Cisco 3600 series routers.

To disable the time-outs call-disconnect timer, set the seconds value to 0.

Use the **timeouts call-disconnect** command to specify the number of seconds the originating end system waits after receiving disconnect before notifying the user to hang up by playing a fast busy tone. During this duration, the user just hears silence. If the command is disabled by setting the value to 0, the user hears silence indefinitely.

Examples

The following example sets a call-disconnect time-out value of 10 seconds on a Cisco 3600 series router voice-port:

```
voice-port 1/0/0
 timeouts call-disconnect 10
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit time-out value for a specified voice port.
timeouts interdigit	Configure the interdigit time-out value for a specified voice port.
timing delay-duration	Configures delay dial signal duration for a specified voice port.

timeouts initial

To configure the initial digit timeout value for a specified voice port, use the **timeouts initial** command in voice-port configuration mode. Use the **no** form of this command to restore the default value.

timeouts initial *seconds*

no timeouts initial *seconds*

Syntax Description

seconds	Initial timeout duration in seconds. Valid entries are any integer from 0 to 120.

Defaults

10 seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **timeouts initial** command to specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits. The timeouts initial timer is activated when the call is accepted and is deactivated when the caller inputs the first digit. If the configured timeout value is exceeded, the caller is notified through the appropriate tone and the call is terminated.

To disable the timeouts initial timer, set the *seconds* value to **0**.

Examples

The following example sets the initial digit timeout value on the Cisco 3600 series to 10 seconds:

```
voice-port 1/0/0
 timeouts initial 10
```

The following example sets the initial digit timeout value on the Cisco MC3810 to 10 seconds:

```
voice-port 1/1
 timeouts initial 10
```

Related Commands

Command	Description
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.

timeouts interdigit

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

timeouts interdigit *seconds*

no timeouts interdigit *seconds*

Syntax Description

seconds	Interdigit timeout duration in seconds. Valid entries are any integer from 0 to 120.

Defaults

10 seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to both the Cisco 3600 series and the Cisco MC3810.

Use the **timeouts interdigit** command to specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The timeouts interdigit timer is activated when the caller inputs a digit and restarted each time the caller inputs another digit until the destination address is identified. If the configured timeout value is exceeded before the destination address is identified, the caller is notified through the appropriate tone and the call is terminated.

To disable the timeouts interdigit timer, set the *seconds* value to **0**.

Examples

The following example sets the interdigit timeout value on the Cisco 3600 series for 10 seconds:

```
voice-port 1/0/0
 timeouts interdigit 10
```

The following example sets the interdigit timeout value on the Cisco MC3810 for 10 seconds:

```
voice-port 1/1
 timeouts interdigit 10
```

Related Commands

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

timeouts wait-release

To configure the timeout value for releasing voice ports on the Cisco MC3810, use the **timeouts wait-release** command in voice-port configuration mode. Use the **no** form of this command to restore the default value.

timeouts wait-release {*value* | **infinity**}

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

Syntax Description

value	The duration in seconds 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. The range is from 5 to 3600 seconds. The default is 30 seconds.
infinity	Indicates that the voice port is never released from call-failure state.

Defaults

30 seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies only to the Cisco MC3810.

Examples

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

```
voice-port 1/1
 timeouts wait-release 180
```

timing clear-wait

To indicate the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port, use the **timing clear-wait** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing clear-wait *milliseconds*

no timing clear-wait *milliseconds*

Syntax Description

milliseconds	Minimum amount of time, in milliseconds, between the inactive seizure signal and the call being cleared. Valid entries on the Cisco 3600 series are numbers from 200 to 2000. Valid entries on the Cisco MC3810 are numbers from 100 to 2000. Supported on E&M ports only.

Defaults

400 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Examples

The following example configures the clear-wait duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing clear-wait 300
```

The following example configures the clear-wait duration on a CiscoMC3810 voice port to 300 milliseconds:

```
voice-port 1/1
 timing clear-wait 300
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing delay-duration

To specify the delay signal duration for a specified voice port, use the **timing delay-duration** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing delay-duration *milliseconds*

no timing delay-duration *milliseconds*

Syntax Description

milliseconds	Delay signal duration for delay dial signalling, in milliseconds. Valid entries are numbers from 100 to 5000. Supported on E&M ports only.

Defaults

2000 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

The call direction for the **timing delay-duration** command is out.

Examples

The following example configures the delay signal duration on a Cisco 3600 series voice port to 3000 milliseconds:

```
voice-port 1/0/0
 timing delay-duration 3000
```

The following example configures the delay signal duration on a Cisco MC3810 voice port to 3000 milliseconds:

```
voice-port 1/1
 timing delay-duration 3000
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing delay-start

To specify the minimum delay time from outgoing seizure to out dial address for a specified voice port, use the **timing delay-start** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing delay-start *milliseconds*

no timing delay-start *milliseconds*

Syntax Description

milliseconds	Minimum delay time, in milliseconds, from outgoing seizure to outdial address. Valid entries are numbers from 20 to 2000. Supported on E&M ports only.

Defaults

300 milliseconds on the Cisco 3600 series.
150 milliseconds on the Cisco MC3810.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

The call direction for the **timing delay-start** command is out.

Examples

The following example configures the delay-start duration on a Cisco 3600 series voice port to 250 milliseconds:

```
voice-port 1/0/0
 timing delay-start 250
```

The following example configures the delay-start duration on a Cisco MC3810 voice port to 250 milliseconds:

```
voice-port 1/1
 timing delay-start 250
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing delay-with-integrity

To specify the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810, use the **timing delay-with-integrity** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing delay-with-integrity *milliseconds*

no delay-with-integrity *milliseconds*

Syntax Description

milliseconds	Duration of the wink pulse for the delay dial, in milliseconds. Valid entries are numbers from 0 to 5000. Supported on E&M ports only.

Defaults

0

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies only to the Cisco MC3810.

Examples

The following example configures the duration of the wink pulse for the delay dial on a Cisco MC3810 voice port to 10 milliseconds:

```
voice-port 1/1
 timing delay-with-integrity 10
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing dial-pulse min-delay

To specify the time between wink-like pulses for a specified voice port, use the **timing dial-pulse min-delay** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing dial-pulse min-delay *milliseconds*

no timing dial-pulse min-delay *milliseconds*

Syntax Description

milliseconds	Time, in milliseconds, between the generation of wink-like pulses. Valid entries are integers from 0 to 5000.

Defaults

300 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

Use the **timing dial-pulse min-delay** command with PBXs requiring a wink-like pulse, even though they have been configured for delay-dial signalling. If the value for this keyword is set to 0, the router will not generate this wink-like pulse. The call signal direction for this command is in.

Examples

The following example configures the time between the generation of wink-like pulses on a Cisco 3600 series voice port to 350 milliseconds:

```
voice-port 1/0/0
 timing dial-pulse min-delay 350
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing dialout-delay

To specify the dial-out delay for the sending digit on a specified voice port on the Cisco MC3810, use the **timing dialout-delay** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing dialout-delay *milliseconds*

no timing dialout-delay *milliseconds*

Syntax Description

milliseconds	Dialout delay, in milliseconds, for the sending digit or cut-through on an FXO trunk or an E&M immediate trunk. Valid entries are from 100 to 5000 milliseconds.

Defaults

300 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies only to the Cisco MC3810.

Examples

The following example configures the dialout delay on a Cisco MC3810 voice port to 350 milliseconds:

```
voice-port 1/1
 timing dialout-delay 350
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing digit

To specify the DTMF digit signal duration for a specified voice port, use the **timing digit** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing digit *milliseconds*

no timing digit *milliseconds*

Syntax Description

milliseconds	The DTMF digit signal duration, in milliseconds. Valid entries are integers from 50 to 100. Supported on FXO, FXS and E&M ports.

Defaults

100 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

The call signal direction for the **timing digit** command is out.

Examples

The following example configures the DTMF digit signal duration on a Cisco 3600 series voice port to 50 milliseconds:

```
voice-port 1/0/0
 timing digit 50
```

The following example configures the DTMF digit signal duration on a Cisco MC3810 voice port to 50 milliseconds:

```
voice-port 1/1
 timing digit 50
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing hookflash-out

To specify the duration of hookflash indications that the gateway generates on an FXO interface, use the **timing hookflash-out** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing hookflash-out *duration*

no timing hookflash-out

Syntax Description

duration	The duration, in milliseconds, of the hookflash. Valid entries are from 50 to 500 milliseconds.

Defaults

300 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(5)T	This command was introduced.

Usage Guidelines

This command specifies a duration of hookflash indications. Hookflash indications may be generated when relayed from an IP network during a VoIP call. Depending on the vendor and country, PBXs and switches vary in their definition of the duration of a hookflash. This command allows you to adjust the hookflash duration appropriately for your network.

Examples

The following example shows how to implement timing for the hookflash with a duration of 200ms after you have configured voice-port 1/0/0:

```
configure terminal
voice-port 1/0/0
 timing hookflash-out 200
```

Related Commands

Command	Description
voice-port	Opens voice-port configuration mode.

timing interdigit

To specify the DTMF interdigit duration for a specified voice port, use the **timing interdigit** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing interdigit *milliseconds*

no timing interdigit *milliseconds*

Syntax Description

milliseconds	DTMF interdigit duration, in milliseconds. Valid entries are numbers from 50 to 500 milliseconds. Supported on FXO, FXS and E&M ports.

Defaults

100 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

The call signal direction for the **timing interdigit** command is out.

Examples

The following example configures the DTMF interdigit duration on a Cisco 3600 series voice port to 150 milliseconds:

```
voice-port 1/0/0
 timing interdigit 150
```

The following example configures the DTMF inter-digit duration on a Cisco MC3810 voice port to 150 milliseconds:

```
voice-port 1/1
 timing interdigit 150
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing percentbreak

To specify the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810, use the **timing percentbreak** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing percentbreak *percent*

no timing percentbreak *percent*

Syntax Description

percent	Percentage of the break period for a dialing pulse. Valid entries are numbers from 20 to 80. Supported on FXO and E&M ports only.

Defaults

50

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to only the Cisco MC3810.

Examples

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

```
voice-port 1/1
 timing percentbreak 30
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing pulse

To specify the pulse dialing rate for a specified voice port, use the **timing pulse** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing pulse *pulses-per-second*

no timing pulse *pulses-per-second*

Syntax Description

pulses-per-second	Pulse dialing rate, in pulses per second. Valid entries are numbers from 10 to 20. Supported on FXO and E&M ports only.

Defaults

20

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

The call signal direction for the **timing pulse** command is out.

Examples

The following example configures the pulse dialing rate on a Cisco 3600 series voice port to 15 pulses per second:

```
voice-port 1/0/0
 timing pulse 15
```

The following example configures the pulse dialing rate on a Cisco MC3810 voice port to 15 pulses per second:

```
voice-port 1/1
 timing pulse 15
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing pulse-interdigit

To specify the pulse interdigit timing for a specified voice port, use the **timing pulse-interdigit** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing pulse-interdigit *milliseconds*

no timing pulse-interdigit *milliseconds*

Syntax Description

milliseconds	Pulse dialing interdigit timing, in milliseconds. Valid entries are integers from 100 to 1000. Supported on FXO and E&M ports only.

Defaults

500 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

The call signal direction for the **timing pulse-interdigit** command is out.

Examples

The following example configures the pulse-dialing interdigit timing on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing pulse-interdigit 300
```

The following example configures the pulse-dialing inter-digit timing on a Cisco MC3810 voice port to 300 milliseconds:

```
voice-port 1/1
 timing pulse-interdigit 300
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing wink-duration

To specify the maximum wink-signal duration for a specified voice port, use the **timing wink-duration** command in voice-port configuration mode. Use the **no** form of this command to restore the default value.

timing wink-duration *milliseconds*

no timing wink-duration *milliseconds*

Syntax Description

milliseconds	Maximum wink-signal duration, in milliseconds, for a wink-start signal. Valid entries are from 100 to 400 milliseconds. Supported on E&M ports only.

Defaults

200 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

The call signal direction for the **timing wink-duration** command is out.

Examples

The following example configures the wink signal duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing wink-duration 300
```

The following example configures the wink signal duration on a Cisco MC3810 voice port to 300 milliseconds:

```
voice-port 1/1
 timing wink-duration 300
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing delay-with-integrity	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing wink-wait

To specify the maximum wink-wait duration for a specified voice port, use the **timing wink-wait** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

timing wink-wait *milliseconds*

no timing wink-wait *milliseconds*

Syntax Description

milliseconds	Maximum wink-wait duration, in milliseconds, for a wink start signal. Valid entries are from 100 to 5000 milliseconds. Supported on E&M ports only.

Defaults

200 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

The call signal direction for the **timing wink-wait** command is out.

Examples

The following example configures the wink-wait duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing wink-wait 300
```

The following example configures the wink-wait duration on a Cisco MC3810 voice port to 300 milliseconds:

```
voice-port 1/1
 timing wink-wait 300
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.

type (settlement)

To point to the specific settlement server, use the **type** command in settlement configuration mode. Use the **no** form of this command to restore the default value.

type *server-type*

Syntax Description

server-type	Indicates the type of the server. In Cisco IOS Release 12.1, only one server type is supported: **osp**.

Defaults

The default is **osp**.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

This command line defines both the Settlement server that is doing the accounting and enables the server to do the accounting. In Cisco IOS Release 12.1, **osp** is the only Settlement server type supported.

Examples

The following example shows how to use the **type** command:

```
settlement 0
 type osp
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of connection attempts to the provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.

Command	Description
settlement	Enters settlement mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

type (voice)

To specify the E&M interface type, use the **type** command in voice-port configuration mode. Use the **no** form of this command to reset the default value.

type {**1** | **2** | **3** | **5**}

no type {**1** | **2** | **3** | **5**}

Syntax Description

1	Indicates the following lead configuration: E—Output, relay to ground. M—Input, referenced to ground.
2	Indicates the following lead configuration: E—Output, relay to SG. M—Input, referenced to ground. SB—Feed for M, connected to –48V. SG—Return for E, galvanically isolated from ground.
3	Indicates the following lead configuration: E—Output, relay to ground. M—Input, referenced to ground. SB—Connected to –48V. SG—Connected to ground.
5	Indicates the following lead configuration: E—Output, relay to ground. M—Input, referenced to –48V.

Defaults

1

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to both the Cisco 3600 series and the Cisco MC3810.

Use the **type** command to specify the E&M interface for a particular voice port. With **1**, the tie-line equipment generates the E-signal to the PBX type grounding the E-lead. The tie-line equipment detects the M-signal by detecting current flow to ground. If you select **1**, a common ground must exist between the line equipment and the PBX.

With **2**, the interface requires no common ground between the equipment, thereby avoiding ground loop noise problems. The E-signal is generated toward the PBX by connecting it to SG. The M-signal is indicated by the PBX connecting it to SB. While Type 2 interfaces do not require a common ground, they do have the tendency to inject noise into the audio paths because they are asymmetrical with respect to the current flow between devices.

> **Note** E&M Type 4 is not a supported option. However, Type 4 operates similarly to Type 2 except for the M-lead operation. On Type 4, the M-lead states are open/ground, compared to Type 2, which is open/battery. Type 4 can interface with Type 2. To use Type 4 you can set the E&M voice port to Type 2 and perform the necessary M-lead rewiring.

With **3**, the interface operates the same as Type 1 interfaces with respect to the E-signal. The M-signal, however, is indicated by the PBX connecting it to SB on assertion and alternately connecting it to SG during inactivity. If you select **3**, a common ground must be shared between equipment.

With **5**, the Type 5 line equipment indicates E-signal to the PBX by grounding the E-lead. The PBX indicates M-signal by grounding the M-lead. A Type 5 interface is quasi-symmetrical in that while the line is up, current flow is more or less equal between the PBX and the line equipment, but noise injection is a problem.

Examples

The following example selects Type 3 as the interface type for your voice port on the Cisco 3600 series:

```
voice-port 1/0/0
 type 3
```

The following example selects Type 3 as the interface type for your voice port on the Cisco MC3810:

```
voice-port 1/1
 type 3
```

unbundle vfc

To unbundle DSPWare from the VCWare and configure the default file and capability lists with default values, use the **unbundle vfc** command in privileged EXEC mode.

unbundle [**high-complexity** | **medium-complexity**] **vfc** *slot-number*

Syntax Description

high-complexity	(Optional) Unbundles the high-complexity firmware set.
slot-number	Indicates VFC slot number.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(2)NA	This command was introduced.
12.0(2)XH	The **high-complexity** and **medium-complexity** keywords were added.

Usage Guidelines

VFCs come with a single bundled image, VCWare, stored in VFC Flash memory. Use the **unbundle vfc** command to unbundle this bundled image into separate files, which are then written to Flash memory. When VCWare is unbundled, it automatically adds DSPWare to Flash memory, creates both the capability and default file lists, and populates these lists with the default files for that version of VCWare. The default file list includes the files that will be used to boot up the system. The capability list defines the available voice codecs for H.323 capability negotiation. These files are used during initial card configuration and for subsequent firmware upgrades.

Before unbundling a VFC software image that you have just copied over to VFC flash, use the **clear vfc** command. Unbundling a DSP firmware set rewrites the default-file and capabilities lists. After unbundling, you must reload the router for any changes to take effect.

Examples

The following example unbundles the high-complexity firmware set into slot 2:

```
router# unbundle high-comp vfc 2
```

Related Commands

Command	Description
copy flash vfc	Copies a new version of VCWare from the Cisco AS5300 motherboard to VFC Flash memory.
copy tftp vfc	Copies a new version of VCWare from a TFTP server to VFC Flash memory.

url

To configure the Internet service provider (ISP) address, use the **url** command in settlement configuration mode. Use the **no** form of this command to restore the default values.

url *url-address*

no url *url-address*

Syntax Description

url-address	Valid URL address is in the following format: http://*fully qualified domain name*[:port]/[URL].

Defaults

No default behavior or values.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced.

Usage Guidelines

You can configure the *url-address* argument multiple times. If you configure multiple URLs for the Settlement server, the gateway attempts to send the request to each URL in the order that you configured these addresses.

Examples

```
settlement 0
 url http://1.2.3.4/
 url http://1.2.3.4:80/
 url https://1.2.3.4:4444/
 url https://yourcompany.com:443/
```

Related Commands

Command	Description
connection-timeout	Configures the time that a connection is maintained after completing a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of connection attempts to the provider.

Command	Description
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement mode and specifies the attributes specific to a settlement provider.
type	Configures an SAA-RTR operation type.

use-proxy

To enable proxy communications for calls between local and remote zones, use the **use-proxy** command in gatekeeper configuration mode. Use the **no** form of this command to either remove a proxy configuration entry for a remote zone or disable proxy communications between local and remote zones.

use-proxy *local-zone-name* {**default** | **remote-zone** *remote-zone-name*} {**inbound-to** | **outbound-from**} {**gateway** | **terminal**}

no use-proxy *local-zone-name* **remote-zone** *remote-zone-name* [{**inbound-to** | **outbound-from**} {**gateway** | **terminal**}]

Syntax Description

local-zone-name	The name or zone name of the gatekeeper, which is usually the fully domain-qualified host name of the gatekeeper. For example, if the domain name is cisco.com, the gatekeeper name might be gk1.cisco.com. However, if the gatekeeper is controlling multiple zones, the name of the gatekeeper for each zone should be a unique string that has a mnemonic value.
default	Defines the default proxy policy for all calls that are not defined by a **use-proxy** command with the **remote-zone** keyword.
remote-zone *remote-zone-name*	Defines a proxy policy for calls to or from a specific remote gatekeeper or zone.
inbound-to	Applies the proxy policy to calls that are inbound to the local zone from a remote zone. Each **use-proxy** command defines the policy for only one direction.
outbound-from	Applies the proxy policy to calls that are outbound from the local zone to a remote zone. Each **use-proxy** command defines the policy for only one direction.
gateway	Defines the type of local device to which the policy applies. The **gateway** option applies the policy only to local gateways.
terminal	Defines the type of local device to which the policy applies. The **terminal** option applies the policy only to local terminals.

Defaults

The local zone uses proxy for both inbound and outbound calls to and from the local H.323 terminals only. Proxy is not used for both inbound and outbound calls to and from local gateways.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
12.0(5)T	This command was introduced.

Usage Guidelines

This command replaces the **zone access** command used in the previous versions of the gatekeeper. When a previous version of gatekeeper is upgraded, any **zone access** commands are translated to **use-proxy** commands. You can use the **show gatekeeper zone status** command to see the gatekeeper proxy configuration.

Examples

In the following example, the local zone sj.xyz.com is configured to use a proxy for inbound calls from remote zones tokyo.xyz.com and milan.xyz.com to gateways in its local zone. The sj.xyz.com zone is also configured to use a proxy for outbound calls from gateways in its local zone to remote zones tokyo.xyz.com and milan.xyz.com:

```
use-proxy sj.xyz.com remote-zone tokyo.xyz.com inbound-to gateway
use-proxy sj.xyz.com remote-zone tokyo.xyz.com outbound-from gateway
use-proxy sj.xyz.com remote-zone milan.xyz.com inbound-to gateway
use-proxy sj.xyz.com remote-zone milan.xyz.com outbound-from gateway
```

Because the default mode disables proxy communications for all gateway calls, only the gateway call scenarios listed above can use the proxy.

In the following example, the local zone sj.xyz.com uses a proxy for only those calls that are outbound from H.323 terminals in its local zone to the specified remote zone germany.xyz.com:

```
no use-proxy sj.xyz.com default outbound-from terminal
use-proxy sj.xyz.com remote-zone germany.xyz.com outbound-from terminal
```

Note that any calls inbound to H.323 terminals in the local zone sj.xyz.com from the remote zone germany.xyz.com use the proxy because the default applies.

The following example shows how to remove one or more proxy statements for the remote zone germany.xyz.com from the proxy configuration list:

```
no use-proxy sj.xyz.com remote-zone germany.xyz.com
```

The command above removes all special proxy configurations for the remote zone germany.xyz.com. After you enter a command like this, all calls between the local zone (sj.xyz.com) and germany.xyz.com are processed according to the defaults defined by any **use-proxy** commands that use the **default** option.

To prohibit proxy use for inbound calls to H.323 terminals in a local zone from a specified remote zone, enter a command similar to the following command:

```
no use-proxy sj.xyz.com remote-zone germany.xyz.com inbound-to terminal
```

This command overrides the default and disables proxy use for inbound calls from remote zone germany.xyz.com to all H.323 terminals in the local zone sj.xyz.com.

Related Commands

Command	Description
show gatekeeper zone status	Displays the status of zones related to a gatekeeper.

vad (dial peer)

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

vad

no vad

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)T	First supported as a dial-peer command on the Cisco MC3810 (in prior releases **vad** was only available as a voice-port command).

Usage Guidelines

Use the **vad** command to enable voice activity detection. With VAD, silence is not sent over the network, only audible speech. If you enable VAD, the sound quality is slightly degraded, but the connection monopolizes much less bandwidth. If you use the **no** form of this command, VAD is disabled and voice data is continuously sent to the IP backbone.

On the Cisco MC3810, VAD can also be assigned to the voice port using the **vad** voice-port configuration command. On the Cisco MC3810, if you enable VAD on the dial peer for Voice over Frame Relay switched calls or permanent calls, the dial peer setting overrides the VAD setting on the voice port.

Note On the Cisco MC3810, the **vad** dial-peer command is enabled by default. The **vad** voice-port command is disabled by default.

Examples

The following example enables VAD for a VoIP dial peer, starting from global configuration mode:

```
dial-peer voice 200 voip
 vad
```

Related Commands

Command	Description
comfort-noise	Generates background noise to fill silent gaps during calls if VAD is activated.
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.

vad (voice-port configuration)

To enable voice activity detection (VAD) for the calls using this voice port, use the **vad** command in voice-port configuration mode. Use the **no** form of this command to disable VAD.

vad

no vad

Syntax Description

This command has no arguments or keywords.

Defaults

VAD is not enabled.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Use the **vad** command to enable voice activity detection. With VAD, silence is not sent over the network, only audible speech. If you enable VAD, the sound quality will be slightly degraded but the connection will monopolize much less bandwidth. If you use the **no** form of this command, VAD is disabled on the voice-port.

Examples

The following example enables VAD:

```
voice-port 1/1
 vad
```

Related Commands

Command	Description
comfort-noise	Generates background noise to fill silent gaps during calls if VAD is activated.
vad (dial-peer configuration)	Enables VAD for the calls using a particular dial peer.

vofr

To enable Voice over Frame Relay (VoFR) on a specific DLCI and to configure specific subchannels on that DLCI, use the **vofr** command in Frame Relay DLCI configuration mode. Use the **no** form of the command to disable VoFR on a specific DLCI.

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

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

Syntax Description

cisco	(Optional) Cisco proprietary voice encapsulation for VoFR with data carried on CID 4 and call-control on CID 5. This option is required on the Cisco MC3810 for applications using switched calls or Cisco trunks.
data	(Optional) Used to select a subchannel (CID) for data other than the default subchannel, which is 4.
cid	(Optional) Specifies the subchannel to be used for data. Valid values are from 4 to 255; the default is 4. If **data** is specified, a valid CID must be entered.
call-control	(Optional) Used to specify that a subchannel will be reserved for call-control signalling. This option is not supported on the Cisco MC3810.
cid	(Optional) Specifies the subchannel to be used for call-control signalling. Valid values are from 4 to 255; the default is 5. If **call-control** is specified and a CID is not entered, the default CID will be used.

Defaults

Disabled

Command Modes

Frame Relay DLCI configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

For switched-vofr calls, use the **vofr cisco** or **vofr call-control** command on the Cisco 2600 series, 3600 series, and 7200 series routers. Switched-vofr calls cannot be made using the **vofr** command by itself, or the **vofr data** *cid* command.

When the **vofr** command is used without the **cisco** keyword, all subchannels on the DLCI are configured for FRF.11 encapsulation. If the **vofr** command is entered without any keywords or arguments, the data subchannel will be CID 4 and there will be no call-control subchannel.

Table 69 describes special conditions and restrictions for the use of the **vofr** command on the Cisco MC3810.

Table 69 Using the vofr Command with the Cisco MC3810

Type of Call	Conditions and Restrictions
FRF.11 trunks	• Do NOT use **cisco** option or **call-control** option. • Use **vofr** or **vofr data** *cid*.
Cisco trunks	Must use **vofr cisco**.
switched-vofr	Must use **vofr cisco**.

Note On the Cisco MC3810 only, the **vofr cisco** command performs the same function as the **frame-relay interface-dlci voice-encap** interface configuration command. Either command is required to enable Voice over Frame Relay. The **vofr cisco** command and the **frame-relay interface-dlci voice-encap** command are mutually exclusive, so you must choose which command to use. The **vofr cisco** command uses weighted fair queueing, which reduces the throughput but provides greater control over the queueing function. The **frame-relay interface-dlci voice-encap** option does not support queueing, which provides greater throughput.

If the **data** keyword is selected, a numeric value must be entered to complete the command. If the **call-control** keyword is selected, you need not enter a numeric value if you wish to accept the default call-control subchannel. See the examples below for clarification.

When the **vofr** command is used on a Cisco MC3810 without the **cisco** keyword, switched calls are not permitted. Only permanent FRF.11-trunk calls can be made.

Note It is not possible to configure the **call-control** keyword on a Cisco MC3810. If this option is configured, the setting is ignored.

Examples

The following example shows how to enable VoFR on Serial 1/1, DLCI 100 on a Cisco 2600 series, 3600 series, or 7200 series router or on an MC3810 concentrator, starting from global configuration mode:

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

The above example configures CID 4 for data; no call-control CID is defined.

The following example configures CID 4 for data, CID 5 for call-control (both defaults):

```
vofr call-control
```

The following example configures CID10 for data, CID 15 for call-control:

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

The following example configures CID 4 for data, CID 15 for call-control:

```
vofr call-control 15
```

The following example configures CID 10 for data, CID 5 for call-control:

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

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

```
vofr data 10
```

The following example configures a Cisco router or MC3810 for a VoFR application with an older release of the MC3810 (prior to Release 12.0(4)T):

```
vofr cisco
```

Related Commands

Command	Description
frame-relay interface-dlci	Assigns a DLCI to a specified Frame Relay subinterface on the router or access server.

voice-card

To configure a voice card and enter voice-card configuration mode, enter the **voice-card** command in global configuration mode.

voice-card *slot*

Syntax Description

slot	A value from 0 to 3 that describes the card location in the module

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	This command was introduced for the Cisco 2600 and 3600 series.

Usage Guidelines

The command is used to enter voice-card configuration mode and set codec complexity.

Examples

The following example enters voice-card configuration mode for the card in slot 1:

```
voice-card 1
```

Related Commands

Command	Description
codec complexity	Specifies call density and codec complexity based on the codec standard you are using.

voice class codec

To create a codec preference list that is independent of a dial peer and can be used on multiple dial peers, use the **voice class codec** command in global configuration mode. Use the **no** form of this command to disable the defined codec preference list.

voice class codec *tag-number*

no voice class codec *tag-number*

Syntax Description

tag-number	Number that identifies a defined codec preference list. Valid entries are from 1 to 10000.

Defaults

No codec preference list defined.

Command Modes

Global configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced.

Examples

The following example creates preference list 99 which can be applied to any dial peer:

```
voice class codec 99
 codec preference 1 g711alaw
 codec preference 2 g711ulaw bytes 80
 codec preference 3 g723ar53
 codec preference 4 g723ar63 bytes 144
 codec preference 5 g723r53
 codec preference 6 g723r63 bytes 120
 codec preference 7 g726r16
 codec preference 8 g726r24
 codec preference 9 g726r32 bytes 80
 codec preference 10 g728
 codec preference 11 g729br8
 codec preference 12 g729r8 bytes 50
```

The following example applies preference list 99 to dial-peer 1919:

```
voice class codec 99
 codec preference 1 g711alaw
 codec preference 2 g711ulaw bytes 80
 codec preference 3 g723ar53
 codec preference 4 g723ar63 bytes 144
 codec preference 5 g723r53
 codec preference 6 g723r63 bytes 120
 codec preference 7 g726r16
 codec preference 8 g726r24
 codec preference 9 g726r32 bytes 80
 codec preference 10 g728
```

```
 codec preference 11 g729br8
 codec preference 12 g729r8 bytes 50
 end
dial-peer voice 1919 voip
 voice-class codec 99
 end
```

Related Commands

Command	Description
codec preference	Specifies a list of preferred codecs to use on a dial peer.

voice class permanent

To create a voice class for a Cisco trunk or FRF.11 trunk, use the **voice class permanent** global configuration command. Use the **no** form of this command to delete the voice class.

voice class permanent *tag*

no voice class permanent *tag*

Syntax Description

tag	Specifies the unique tag number you assign to the permanent voice class. The valid range for this tag is 1 to 10000. The *tag* number must be unique on the router.

Defaults

No voice class is configured.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced.

Usage Guidelines

This command can be used for VoFR, VoATM, and VoHDLC trunks.

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

Examples

The following example shows how to create a permanent voice class starting from global configuration mode:

```
voice class permanent 10
```

Related Commands

Command	Description
signal keepalive	Configures the keepalive signalling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of the call.
signal timing oos	Configures the signal timing parameter for the OOS state of the call.
voice class permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

voice-class permanent

To assign a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer, use the **voice-class permanent** command in dial-peer configuration mode. Use the **no** form of this command to remove the voice-class assignment from the dial peer.

voice-class permanent *tag-number*

no voice-class permanent *tag-number*

Syntax Description

tag-number	Specifies the unique tag number assigned to the permanent voice class. The valid range for this tag is 1 to 10000. The *tag* number maps to the tag number created using the **voice class permanent** global configuration command.

Defaults

This command has no default.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(4)T	This command was introduced.

Usage Guidelines

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

Examples

The following example shows how to configure a permanent voice class starting from global configuration mode, configure parameters for that voice class, and then assign the voice class to a dial peer:

```
voice class permanent 10
 signal keepalive 3
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
signal keepalive	Configures the keepalive signalling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of the call.
signal timing oos	Configures the signal timing parameter for the OOS state of the call.
signal-type	Sets the signalling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.

voice confirmation-tone

To disable the two-beep confirmation tone for PLAR or PLAR off premises extension (OPX) connections, use the **voice confirmation-tone** command in voice-port configuration mode. Use the **no** form of this command to enable the two-beep confirmation tone.

voice confirmation-tone

no voice confirmation-tone

Syntax Description

This command has no arguments or keywords.

Defaults

The two-beep confirmation tone is heard on the PLAR or PLAR OPX connection.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies only to the Cisco MC3810.

Use this command to disable the two-beep confirmation tone that a caller hears when picking up the handset for PLAR and PLAR OPX connections. This command is only valid if the voice port **connection** command is set to PLAR or PLAR OPX.

Examples

The following example disables the two-beep confirmation tone on voice port 1/1 on the Cisco MC3810:

```
voice-port 1/1
 connection plar-opx
 voice confirmation-tone
```

Related Commands

Command	Description
connection	Specifies a connection mode for a voice port.

voice-encap

To define the data segmentation size on an HDLC interface to support Voice over HDLC, use the **voice-encap** command in interface configuration mode. Use the **no** form of this command to delete the setting.

voice-encap *size*

no voice-encap *size*

Syntax Description

size	The size of the data segmentation. The valid range is from 8 to 1600.

Defaults

No data segmentation size is defined.

Command Modes

Interface configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over HDLC on the Cisco MC3810.

Examples

The following example configures serial interface 1 to support a data segmentation size of 64 for Voice over HDLC:

```
interface serial 1
 voice-encap 64
```

voice-group

j,To configure a list of timeslots for voice channel-associated signalling (CAS) or common channel signalling (CCS) on the T1/E1 controller, use the **voice-group** command in controller configuration mode. Use the **no** form of this command to delete the CAS group.

voice-group *channel-no* **timeslots** *timeslot-list* **type** {**e&m-immediate** | **e&m-delay** | **e&m-wink** | **e&m-melcas** | **fxs-ground-start** | **fxs-loop-start** | **fxs-melcas** | **fxo-ground-start** | **fxo-loop-start** | **fxo-melcas**} | {**ext-sig-master** | **ext-sig-slave**}

no voice-group *channel-no* **timeslots** *timeslot-list* **type** {**e&m-immediate** | **e&m-delay** | **e&m-wink** | **e&m-melcas** | **fxs-ground-start** | **fxs-loop-start** | **fxs-melcas** | **fxo-ground-start** | **fxo-loop-start** | **fxo-melcas**} | {**ext-sig-master** | **ext-sig-slave**}

Syntax Description

channel-no	Channel number to identify the CAS group. The valid range is from 0 to 23.
timeslots *timeslot-list*	A list of timeslots which comprise the CAS group. The valid range is from 1-24 for T1, and from 1-15 and 17-31 for E1.
type	The type of voice interface. Choose one of the following **type** options: The following **type** options are available if the **mode cas** command is enabled: • **e&m-immediate**—E&M immediate. • **e&m-delay**—E&M delay. • **e&m wink**—E&M wink. • **e&m melcas**—E&M mercury exchange limited channel-associated signalling (MEL CAS). • **fxs-ground-start**—Foreign Exchange Station ground-start. • **fxs-loop-start**—Foreign Exchange Station loop start. • **fxs-melcas**—Foreign Exchange Station MELCAS. • **fxo-ground-start**—Foreign Exchange Office ground start. • **fxo-loop-start**—Foreign Exchange Office loop start. • **fxo-melcas**—Foreign Exchange Office MELCAS. The following **type** options are available only if the **mode ccs** command is enabled: • **ext-sig-master**—Specifies the channel to automatically generate the off-hook signal and stay in the off-hook state. • **ext-sig-slave**—Specifies the channel to automatically generate the answer signal when a call is terminated to that channel.

Defaults

No default behavior or values.

Command Modes

Controller configuration

Command History

Release	Modification
11.3 MA	This command was introduced.
12.0(2)T	The **ext-sig-master** and **ext-sig-slave** keywords were added.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

This command is only available if the **mode cas** or **mode ccs** command is enabled.

Note Channel groups, CAS voice groups, and TDM groups all use group numbers. All group numbers configured for channel groups, CAS voice groups, and TDM groups must be unique on the local Cisco MC3810 concentrator. For example, you cannot use the same group number for a channel group and for a TDM group.

Examples

The following example configures a voice group on controller T1 0 on a Cisco MC3810:

```
controller T1 0
 mode cas
 voice-group 10 timeslots10 64
```

Related Commands

Command	Description
channel-group	Defines the timeslots that belong to each T1 or E1 circuit.
tdm-group	Configures a list of timeslots for creating clear channel groups (pass-through) for TDM cross-connect.

voice hunt user-busy

To configure a tandem router so it does not stop dial-peer hunting if it receives a user-busy disconnect code from a destination router, use the **voice hunt user-busy** command in global configuration mode. To configure the tandem router so it does stop dial-peer hunting if it receives a user-busy disconnect code (the default option), use the **no** form of this command.

voice hunt user-busy

no voice hunt user-busy

Syntax Description

This command has no arguments or keywords.

Defaults

The tandem router stops dial-peer hunting when it receives a user-busy disconnect code.

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)T	This command was introduced.

Usage Guidelines

This command applies only to routers acting as tandem nodes in a Voice over Frame Relay environment.

This command is used for a configuration in which a tandem router is configured with multiple dial peer entries that route a call to the same destination number, but on different destination routers. In this configuration, after all routes to the first router entry in the dial-peer list are active, a new call will not "roll over" to the next router in the dial-peer list.

This failure to route to the second destination router happens when the bandwidth on the Voice over Frame Relay interface is greater than the maximum capacity of the first destination router. This condition allows the tandem router to attempt to place a new call to the first router because it has indications from the first router that there is more capacity based on the bandwidth setting. When the first destination router receives the call, if all of the ports are in use, the destination router returns a "user-busy" disconnect reason code to the tandem router. The tandem router interprets the disconnect reason code as meaning there is no available destination for the call, causing it to return a busy tone to the initiating caller

The tandem router fails to try other routers in the dial-peer list after receiving a "user disconnect" reason code, and so it terminates the call attempt. Using this command, you can perform dial-peer hunting on multiple destination routers even if the tandem router receives a "user-busy" disconnect reason code from one of the destination routers.

Examples

The following example configures the tandem router to continue dial-peer hunting if it receives a "user-busy" disconnect code from a destination router:

```
voice hunt user-busy
```

Related Commands

Command	Description
preference	Indicates the preferred order of a dial peer within a rotary hunt group.

voice local-bypass

To directly cross-connect local POTS calls without going through a Digital Signal Processor (DSP), use the **voice local-bypass** command in global configuration mode. Use the **no** form of this command to cancel the voice local-bypass disable operation.

voice local-bypass

no voice local-bypass

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Global configuration

Command History

Release	Modification
11.3 MA	This command was introduced.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

This command allows you to pass uncompressed voice traffic for local POTS calls.

Examples

The following example configures the Cisco MC3810 to directly cross-connect local calls without going through a DSP:

```
voice local-bypass
```

voice-port

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

Cisco 1750 router

voice-port *slot-number/port*

Cisco 2600 and Cisco 3600 series router

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

Cisco MC3810

voice-port *slot/port*

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**}

Cisco 7200 series router

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

Cisco uBR924 cable access router

voice-port *number*

Syntax Description

Cisco 1750 router

slot-number	Slot number in the router where the VIC is installed. Valid entries are from 0 to 2, depending on the slot where it has been installed.
port	Indicates the voice port. Valid entries are 0 or 1.

Cisco 2600 and Cisco 3600 series routers

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.
slot	The router location where the voice port adapter is installed. Valid entries are from 0 to 3.

port	Indicates the voice interface card location. Valid entries are 0 or 3.
dso-group-no	Indicates the defines DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

Cisco MC3810:

slot/port	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.

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.

For the Cisco 7200 series router

slot	The router location where the voice port adapter is installed. Valid entries are from 0 to 3.
port	Indicates the voice interface card location. Valid entries are 0 or 1.
dso-group-no	Indicates the defines DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.
slot-number	Indicates the 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	Indicates the subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1.
port	Indicates the voice port number. Valid entries are 0 or 1.

For the Cisco uBR924 cable access router

number	Indicates the RJ-11 connectors installed in the Cisco uBR924. Valid entries are 0 (which corresponds to the RJ-11 connector labeled V1) and 1 (which corresponds to the RJ-11 connector labeled V2.

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 universal access server, the Cisco 7200 series router, and the Cisco 1750 router were added. Arguments for the Cisco 2600 and Cisco 3600 series router were 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, defines the type of dial peer, and defines the tag number associated with a dial peer.

voice-port busyout

To place all voice ports associated with a serial or ATM interface into a busyout state, use the **voice-port busyout** command in interface configuration mode. To remove the busyout state on the voice ports associated with this interface, use the **no** form of this command.

voice-port busyout

no voice-port busyout

Syntax Description

This command has no arguments or keywords.

Defaults

The voice port(s) on the interface are not in busyout state.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(3)T	This command was introduced.

Usage Guidelines

This command is only supported on the Cisco MC3810.

This command busies out all voice port associated with the interface, except any voice ports configured to busy out under specific conditions using the **busyout monitor** and **busyout-seized** commands.

Examples

The following example turns the voice-port(s) associated with serial interface 1 into busyout state:

```
interface serial 1
 voice-port busyout
```

The following example turns the voice-port(s) associated with ATM interface 0 into busyout state:

```
interface atm 0
 voice-port busyout
```

Related Commands

Command	Description
busyout forced	Forces a voice port on the Cisco MC3810 multiservice concentrator into the busyout state.
busyout-monitor	Places a voice port on the Cisco MC3810 multiservice concentrator into the busyout monitor state.
busyout-seize	Changes the busyout seize procedure fro a voice port on the Cisco MC3810 multiservice concentrator.
show voice busyout	Displays information about the voice busyout state on the Cisco MC3810 multiservice concentrator.

zone access

To configure the accessibility of your localzone zone, use the **uuuuuuuuzone access** command in gatekeeper configuration mode. To remove any accessibility configurations, use the **no** form of this command.

zone access *local-zone-name* {**default** | **remote-zone** *remote-zone-name*} {**direct** | **proxied**}

no zone access *local-zone-name* **remote-zone** *remote-zone-name*

Syntax Description

local-zone-name	Name of local zone (synonymous with local gatekeeper).
default	Use with the **direct** or **proxied** keyword to define the mode of behavior for all remote zones that have not been specially named using the **remote-zone** *remote-zone-name* keyword and argument combination.
remote-zone *remote-zone-name*	Name of remote zone (synonymous with remote gatekeeper) for which a special mode of behavior is defined.
direct	Configures direct calls (without use of proxies) between endpoints. The local zone (or gatekeeper) offers the local endpoint IP address instead of a local proxy's IP address.
proxied	Configures calls using proxies between endpoints. The local zone (or gatekeeper) offers the local proxy's IP address instead of the local endpoint's address.

Defaults

The local zone allows proxied access for all remote zones.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced.

Usage Guidelines

By default, a gatekeeper will offer a local proxy IP address when queried by a remote gatekeeper about a target local endpoint. This is considered proxied access. By using the **zone access** command, you can configure the local gatekeeper to offer the local endpoint address instead of the local proxy address. This is considered direct access.

Note The **zone access** command, configured on your local gatekeeper, only affects the use of proxies for incoming calls (that is, it does not affect the use of local proxies for outbound calls). When originating a call, a gatekeeper will use a proxy only if the remote gatekeeper offers a proxy at the remote end. A call between two endpoints in the same zone will always be a direct (nonproxied) call.

You can define the accessibility behavior of a local zone relative to certain remote zones using the **remote-zone** *remote-zone-name* keyword and argument combination with the **direct** or **proxied** keyword. You can define the default behavior of a local zone relative to all other remote zones using the **default** keyword with the **direct** or **proxied** keywords. To remove an explicitly named remote zone so that it is governed by the default-behavior rule, use the **no zone access** command.

Examples

The following example allows direct access to the local zone eng.xyz.com from remote zones within xyz corporation. All other remote locations will have proxied access to *eng.xzy.com*.

```
zone local eng.xyz.com xyz.com
zone access eng.xyz.com remote-zone mfg.xyz.com direct
zone access eng.xyz.com remote-zone mktg.xyz.com direct
zone access eng.xyz.com remote-zone sales.xyz.com direct
zone access eng.xyz.com default proxied
```

The following example supposes that only local gatekeepers within xyz.com have direct access to each other because your corporation has firewalls or you do not advertise your gatekeepers externally. You have excellent QoS within your corporate network, except for a couple of foreign offices. In this case, use proxies with the foreign offices (in Milan and Tokyo) and nowhere else.

```
zone local sanjose.xyz.com xyz.com
zone access sanjose.xyz.com default direct
zone access sanjose.xyz.com remote-zone milan.xyz.com proxied
zone access sanjose.xyz.com remote-zone tokyo.xyz.com proxied
```

Related Commands

Command	Description
show proxy h323 calls	Displays a list of each active call on the proxy.
zone local	Specifies a zone controlled by a gatekeeper.

zone bw

To set the maximum bandwidth allowed in a gatekeeper zone at any one time, use the **zone bw** command in gatekeeper configuration mode. To remove the maximum bandwidth setting and make the bandwidth unlimited, use the **no** form of this command.

zone bw *gatekeeper-name max-bandwidth*

no zone bw *gatekeeper-name max-bandwidth*

Syntax Description

gatekeeper-name	Name of the gatekeeper controlling the zone.
max-bandwidth	Maximum bidirectional bandwidth in kilobits per second (kbps) allowed in the zone at any one time.

Defaults

Bandwidth is unlimited.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced.

Examples

The following example sets the maximum bandwidth to 1000 kbps for zone gk1:

```
zone bw gk1 1000
```

Related Commands

Command	Description
show proxy h323 calls	Displays a list of each active call on the proxy.

zone local

To specify a zone controlled by a gatekeeper, use the **zone local** command in gatekeeper configuration mode. Use the **no** form of this command to remove a zone controlled by a gatekeeper.

zone local *gatekeeper-name domain-name* [*rasIPaddress*]

no zone local *gatekeeper-name domain-name*

Syntax Description

gatekeeper-name	The gatekeeper's name or zone name. This is usually the fully domain-qualified host name of the gatekeeper. For example, if the *domain-name* is cisco.com, the *gatekeeper-name* might be *gk1.cisco.com.* However, if the gatekeeper is controlling multiple zones, the *gatekeeper-name* for each zone should be some unique string that has a mnemonic value.
domain-name	The domain name served by this gatekeeper.
rasIPaddress	(Optional) The IP address of one of the interfaces on the gatekeeper. When the gatekeeper responds to gatekeeper discovery messages, it signals the endpoint or gateway to use this address in future communications. **Note** Setting this address for one local zone makes it the address used for all local zones.

Defaults

No local zone is defined.

Note The gatekeeper cannot operate without at least one local zone definition. Without local zones, the gatekeeper goes to an inactive state when the **no shutdown** command is issued.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced.

Usage Guidelines

Multiple local zones can be defined. The gatekeeper manages all configured local zones. Intra-zone and inter-zone behavior remains the same (zones are controlled by the same or different gatekeepers).

Only one *rasIPaddress* argument can be defined for all local zones. You cannot configure each zone to use a different RAS IP address. If you define this in the first zone definition, you can omit it for all subsequent zones, which automatically pick up this address. If you set it in a subsequent **zone local** command, it changes the RAS address of all previously configured local zones as well. Once defined, you can change it by re-issuing any **zone local** command with a different *rasIPaddress* argument.

If the *rasIPaddress* argument is an HSRP virtual address, it automatically puts the gatekeeper into HSRP mode. In this mode, the gatekeeper assumes STANDBY or ACTIVE status according to whether the HSRP interface is on STANDBY or ACTIVE status.

You cannot remove a local zone if there are endpoints or gateways registered in it. To remove the local zone, shut down the gatekeeper first, which forces unregistration.

Multiple zones are controlled by multiple logical gatekeepers on the same Cisco IOS platform.

The maximum number of local zones defined in a gatekeeper should not exceed 100.

This command can also be used to change the IP address used by the gatekeeper.

Examples

The following example creates a zone controlled by a gatekeeper in the domain called cisco.com:

```
zone local gk1.cisco.com cisco.com
```

Related Commands

Command	Description
show proxy h323 calls	Displays a list of each active call on the proxy.
zone subnet	Specifies a zone controlled by a gatekeeper.

zone prefix

To add a prefix to the gatekeeper zone list, use the **zone prefix** command in gatekeeper configuration mode. To remove knowledge of a zone prefix, use the **no** form of this command with the gatekeeper name and prefix. To remove the priority assignment for a specific gateway, use the **no** form of this command with the **gw-priority** option.

zone prefix *gatekeeper-name e164-prefix* [**gw-priority** *pri-0-to-10 gw-alias* [*gw-alias*, ...]]

no zone prefix *gatekeeper-name e164-prefix* [**gw-priority** *pri-0-to-10 gw-alias* [*gw-alias*, ...]]

Syntax Description

gatekeeper-name	The name of a local or remote gatekeeper, which must have been defined by using the **zone local** or **zone remote** command.
e164-prefix	An E.164 prefix in standard form followed by dots (.). Each dot represent a number in the E.164 address. For example, 212....... is matched by 212 and any seven numbers. **Note** Although a dot representing each digit in an E.164 address is the preferred configuration method, you can also enter an asterisk (*) to match any number of digits.
gw-priority *pri-0-to-10 gw-alias*	(Optional) Use the **gw-priority** option to define how the gatekeeper selects gateways in its local zone for calls to numbers beginning with prefix *e164-prefix*. Do not use this option to set priority levels for a prefix assigned to a remote gatekeeper. Use values from 0 to 10. A 0 value prevents the gatekeeper from using the gateway *gw-alias* for that prefix. Value 10 places the highest priority on gateway *gw-alias*. If you do not specify a priority value for a gateway, the value 5 is assigned. To assign the same priority value for one prefix to multiple gateways, list all the gateway names after the *pri-0-to-10* value. The *gw-alias* name is the H.323 ID of a gateway that is registered or will register with the gatekeeper. This name is set on the gateway with the **h323-gateway voip h.323-id** command.

Defaults

No knowledge of its own prefix or the prefix of any other zone is defined.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(6)Q	This command was introduced.
11.3(7)NA	This command was modified for H.323 Version 1.
12.0(5)T	This display format was modified for H.323 Version 2.

Usage Guidelines

A gatekeeper can handle more than one zone prefix, but a zone prefix cannot be shared by more than one gatekeeper. If you have defined a zone prefix as being handled by a gatekeeper and now define it as being handled by a second gatekeeper, the second assignment cancels the first.

If you need a gatekeeper to handle more than one prefix, but for cost reasons you want to be able to group its gateways by prefix usage, there are two ways to do it.

The first method is simpler, has less overhead, and is recommended if your gateways can be divided into distinct groups, where each group is to be used for a different set of prefixes. For instance, if a group of gateways is used for calling area codes 408 and 650, and another group is used for calling area code 415, you can use this method. In this case, you define a local zone for each set of prefixes, and have the group of gateways to be used for that set of prefixes register with that specific local zone. Do not define any gateway priorities. All gateways in each local zone are treated equally in the selection process.

However, if your gateways cannot be cleanly divided into nonintersecting groups (for instance if one gateway is used for calls to 408 and 415 and another gateway is used for calls to 415 and 650, and so on), you can put all these gateways in the same local zone and use the **gw-priority** option to define which gateways will be used for which prefixes.

When choosing a gateway, the gatekeeper first looks for the longest zone prefix match; then it uses the priority and the gateway status to select from the gateways. If all gateways are available, the gatekeeper chooses the highest priority gateway. If all the highest priority gateways are busy (see the gateway **resource threshold** command), a lower priority gateway is selected.

Note The **zone prefix** command matches a prefix to a gateway. It does not register the gateway. The gateway must register with the gatekeeper before calls can be completed through that gateway.

Examples

The following example shows how you can define multiple local zones for separating your gateways:

```
zone local gk408or650 xyz.com
zone local gk415 xyz.com
zone prefix gk408or650 408.......
zone prefix gk408or650 650.......
zone prefix gk415 415.......
```

Now you need to configure all the gateways to be used for area codes 408 or 650 to register with gk408or650 and all gateways to be used for area code 415 to register with gk415. On Cisco voice gateways, you configure the gateways to register with the appropriate gatekeepers by using the **h323 voip id** command.

The following example shows how you can put all your gateways in the same zone but use the **gw-priority** keyword to determine which gateways will be used for calling different area codes:

```
zone local localgk xyz.com
zone prefix localgk 408.......
zone prefix localgk 415....... gw-pri 10 gw1 gw2
zone prefix localgk 650....... gw-pri 0 gw1
```

The commands shown accomplish the following tasks:

- Domain xyz.com is assigned to gatekeeper localgk.
- Prefix 408...... is assigned to gatekeeper localgk, and no gateway priorities are defined for it; therefore, all gateways registering to localgk can be used equally for calls to the 408 area code. No special gateway lists are built for the 408....... prefix; selection is made from the master list for the zone.

- The prefix 415....... is added to gatekeeper localgk, and priority 10 is assigned to gateways gw1 and gw2.
- Prefix 650....... is added to gatekeeper localgk, and priority 0 is assigned to gateway gw1.

A priority 0 is assigned to gateway gw1 to exclude it from the gateway pool for prefix 650........ When gateway gw2 registers with gatekeeper localgk, it is added to the gateway pool for each prefix as follows:

- For gateway pool for 415......., gateway gw2 is set to priority 10.
- For gateway pool for 650......., gateway gw2 is set to priority 5.

The following example changes gateway gw2 from priority 10 for zone 415....... to the default priority 5:

```
no zone prefix localgk 415....... gw-pri 10 gw2
```

The following example changes both gateways gw1 and gw2 from priority 10 for zone 415....... to the default priority 5:

```
no zone prefix localgk 415....... gw-pri 10 gw1 gw2
```

In the preceding example, the prefix 415....... remains assigned to gatekeeper localgk. All gateways that do not specify a priority level for this prefix are assigned a default priority of 5. The following example removes the prefix and all associated gateways and priorities from this gatekeeper:

```
no zone prefix localgk 415.......
```

Related Commands

Command	Description
resource threshold	Configures a gateway to report H.323 resource availability to the gatekeeper of the gateway.
register	Configures a gateway to register or deregister a fully qualified dial-peer E.164 address with a gatekeeper.
show call resource voice threshold	Displays the threshold configuration settings and status for an H.323 gateway.
show gateway	Displays the current gateway status.

zone remote

To statically specify a remote zone if domain name service (DNS) is unavailable or undesirable, use the **zone remote** command in gatekeeper configuration mode. To remove the remote zone, use the **no** form of this command.

zone remote *other-gatekeeper-name other-domain-name other-gatekeeper-ip-address* [*port-number*]

no zone remote *other-gatekeeper-name other-domain-name other-gatekeeper-ip-address* [*port-number*]

Syntax Description

other-gatekeeper-name	Name of the remote gatekeeper.
other-domain-name	Domain name of the remote gatekeeper.
other-gatekeeper-ip-address	IP address of the remote gatekeeper.
port-number	(Optional) RAS signalling port number for the remote zone. Value ranges from 1 to 65535. If this is not set, the default is the well-known RAS port number 1719.

Defaults

No remote zone is defined. DNS will locate the remote zone.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Usage Guidelines

All gatekeepers do not have to be in DNS. For those that are not, use the **zone remote** command so that the local gatekeeper knows how to access them. In addition, you may wish to improve call response time slightly for frequently accessed zones. If the **zone remote** command is configured for a particular zone, you do not need to make a DNS lookup transaction.

The maximum number of zones defined on a gatekeeper varies depending on the mode or the call model or both. For example, a directory gatekeeper may be in the mode of being responsible for forwarding LRQs and not handling any local registrations and calls; The call model might be E.164 addressed calls instead of H.323-ID addressed calls.

For a directory gatekeeper that does not handle local registrations and calls, the maximum remote zones defined should not exceed 10,000; An additional 4 MB of memory is required to store this maximum number of remote zones.

For a gatekeeper that handles local registrations and only E.164 addressed calls, the number of remote zones defined should not exceed 2000.

For a gatekeeper that handles H.323-ID calls, the number of remote zones defined should not exceed 200.

Examples

The following example configures the local gatekeeper to reach targets of the form *xxx*.cisco.com by sending queries to the gatekeeper named sj3.cisco.com at IP address 1.2.3.4:

```
zone remote sj3.cisco.com cisco.com 1.2.3.4
```

Related Commands

Command	Description
show proxy h323 calls	Displays a list of each active call on the proxy.
zone local	Specifies a zone controlled by a gatekeeper.

zone subnet

To configure a gatekeeper to accept discovery and registration messages sent by endpoints in designated subnets, use the **zone subnet** command in gatekeeper configuration mode. To disable the gatekeeper from acknowledging discovery and registration messages from subnets or remove subnets entirely, use the **no** form of this command.

zone subnet *local-gatekeeper-name* {**default** | *subnet-address* {*/bits-in-mask* | *mask-address*}} **enable**

no zone subnet *local-gatekeeper-name* {**default** | *subnet-address* {*/bits-in-mask* | *mask-address*}} **enable**

Syntax Description

local-gatekeeper-name	Name of the local gatekeeper.
default	Applies to all other subnets that are not specifically defined by the **zone subnet** command.
subnet-address	Address of the subnet being defined.
/bits-in-mask	Number of bits of the mask to be applied to the subnet address.
mask-address	Mask (in dotted string format) to be applied to the subnet address.
enable	Gatekeeper accepts discovery and registration from the specified subnets.

Defaults

The local gatekeeper accepts discovery and registration requests from all subnets. If the request specifies a gatekeeper name, it must match the local gatekeeper name or the request will not be accepted.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA and 12.0(3)T	This command was introduced.

Usage Guidelines

You can use the **zone subnet** command more than once to create a list of subnets controlled by a gatekeeper. The subnet masks do not have to match actual subnets in use at your site. For example, to specify a particular endpoint, you can supply its address with a 32-bit netmask.

Examples

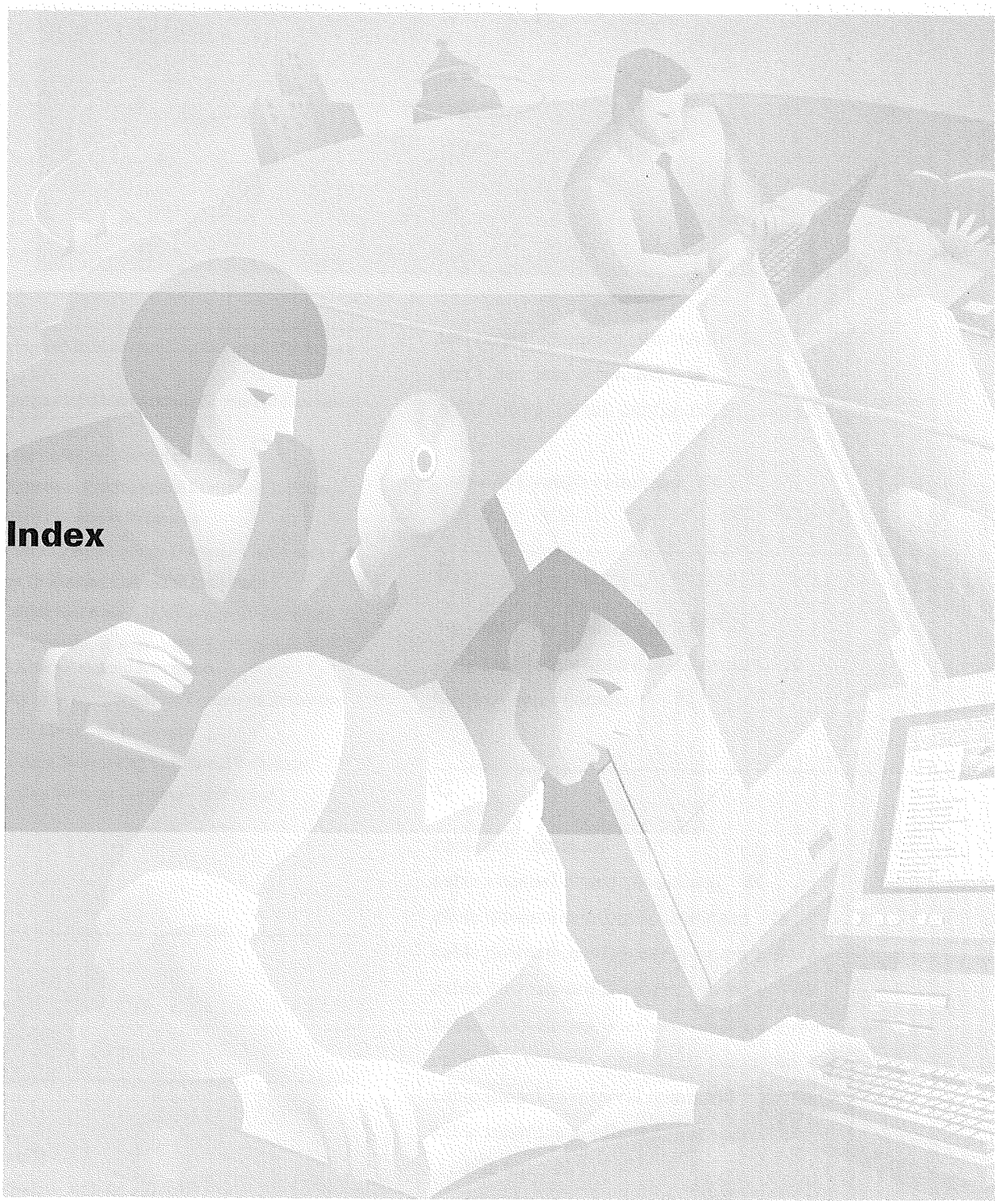
The following example starts by disabling the gatekeeper, gk1.cisco.com, from accepting discovery and registration messages from all subnets. Next, gk1.cisco.com is configured to accept discovery and registration messages from all H.323 nodes on the subnet 172.21.127.0. In addition, gk1.cisco.com is configured to accept discovery and registration messages from a particular endpoint with the IP address 172.21.128.56.

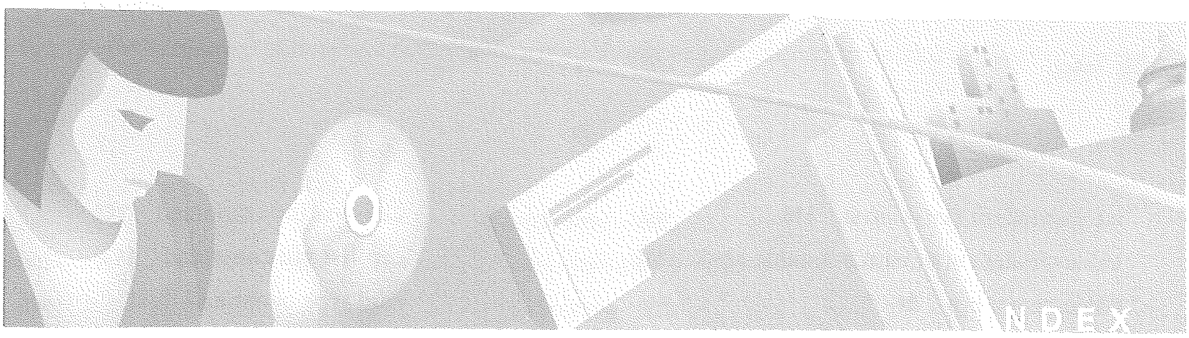
```
no zone subnet gk1.cisco.com default enable
zone subnet gk1.cisco.com 172.21.127.0/24 enable
zone subnet gk1.cisco.com 172.21.128.56/32 enable
```

Related Commands

Command	Description
show gatekeeper zone status	Displays the status of zones related to a gatekeeper.
zone local	Specifies a zone controlled by a gatekeeper.

Index





B1R	Cisco IOS Bridging and IBM Networking Command Reference, Volume I
B2R	Cisco IOS Bridging and IBM Networking Command Reference, Volume II
DR	Cisco IOS Dial Services Command Reference
FR	Cisco IOS Configuration Fundamentals Command Reference
IR	Cisco IOS Interface Command Reference
MR	Cisco IOS Multiservice Applications Command Reference
P1R	Cisco IOS IP and IP Routing Command Reference
P2R	Cisco IOS AppleTalk and Novell IPX Command Reference
P3R	Cisco IOS Apollo Domain, Banyan VINES, DECnet, ISO CLNS, and XNS Command Reference
QR	Cisco IOS Quality of Service Solutions Command Reference
SR	Cisco IOS Security Command Reference
WR	Cisco IOS Wide-Area Networking Command Reference
XR	Cisco IOS Switching Services Command Reference

Symbols

A

B

C

D

E

F

G

H

I

L

M

N

O

P

Q

R

S

Z

Cisco Systems

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
Cisco Systems Europe s.a.r.l.
Parc Evolic, Batiment L1/L2
16 Avenue du Quebec
Villebon, BP 706
91961 Courtaboeuf Cedex
France
http://www-europe.cisco.com
Tel: 33 1 69 18 61 00
Fax: 33 1 69 28 83 26

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Headquarters
Nihon Cisco Systems K.K.
Fuji Building, 9th Floor
3-2-3 Marunouchi
Chiyoda-ku, Tokyo 100
Japan
http://www.cisco.com
Tel: 81 3 5219 6250
Fax: 81 3 5219 6001

Cisco Systems has more than 200 offices in the following countries. Addresses, phone numbers, and fax numbers are listed on the
Cisco Connection Online Web site at http://www.cisco.com/offices.

Argentina • Australia • Austria • Belgium • Brazil • Canada • Chile • China • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE
Finland • France • Germany • Greece • Hong Kong • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia
Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Singapore
Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey • Ukraine • United Kingdom • United States • Venezuela

Printed in the USA on recycled paper containing 10% postconsumer waste.

Cover Part Number: 56-2343-06

78-10259-01