CHAPTER 4

VoIP Commands

This chapter provides an alphabetical listing of all the VoIP commands that are new or specific to the Cisco 1750 router. All other commands used with this feature are documented in the Cisco IOS Release 12.0T command reference documents.

Table 4-1 lists and describes the commands in this chapter that are used to configure and monitor VoIP.

Table 4-1 Commands Used to Configure and Monitor VoIP

Command	Description
acc-qos	Generate an SNMP event if the QoS drops below a specified level.
answer-address	Specify the full E.164 telephone number to identify the dial peer of an incoming call.
codec	Specify the voice coder rate of speech for a dial peer.
comfort-noise	Specify whether or not background noise should be generated.
connection	Specify a connection mode for a specified voice port.
cptone	Configure a voice call progress tone locale.
description	Include a description of what this voice port is connected to.
destination-pattern	Specify either the prefix or the full E.164 telephone number to be used for a dial peer.
dial-control-mib	Specify attributes for the call history table.
dial-peer voice	Enter the dial peer configuration mode.
dial-type	Specify the type of out-dialing for voice-port interfaces.
echo-cancel coverage	Adjust the size of the echo cancel.
echo-cancel enable	Enable the echo cancel feature.
expect-factor	Specify when the router will generate an alarm to the network manager.
fax-rate	Establish the rate at which a fax is sent to the specified dial peer.
icpif	Specify the Calculated Planning Impairment Factor (ICPIF) for calls sent by a dial peer.
impedance	Specify the terminating impedance of a voice-port interface.
input gain	Configure a specific input gain value.
ip precedence	Set IP precedence (priority) for packets sent by the dial peer.
ip udp checksum	Calculate the UDP checksum for voice packets transmitted by the dial peer.
music-threshold	Specify the threshold for on-hold music for a specified voice port.
non-linear	Enable nonlinear processing in the echo canceller.

Table 4-1 Commands Used to Configure and Monitor VoIP

Command	Description
num-exp	Define how to expand an extension number into a particular destination pattern.
operation	Select a specific cabling scheme for E&M ports.
output attenuation	Configure a specific output attenuation value.
port	Associate a dial peer with a specific voice port.
prefix	Specify the prefix of the dialed digits for this dial peer.
req-qos	Specify the desired QoS to be used in reaching a specified dial peer.
ring frequency	Specify the ring frequency for a specified FXS voice port.
ring number	Specify the number of rings for a specified FXO voice port.
session protocol	Establish a session protocol for calls between the local and remote routers .
session target	Specify a network-specific address for a specified dial peer.
show call active voice	Show the active call table.
show call history voice	Display the call-history table.
show controllers voice	Display information about voice related hardware.
show diag	Display hardware information for the router.
show dial-peer voice	Display configuration information for dial peers.
show dialplan incall number	Pair different voice ports and telephone numbers together for troubleshooting.
show dialplan number	Show which dial peer is reached when a particular telephone number is dialed.
show num-exp	Show the number expansions configured.
show voice port	Display configuration information about a specific voice port.
shutdown (dial-peer configuration)	Change the administrative state of the selected dial peer from up to down.
shutdown (voice-port configuration)	Take the voice ports for a specific VIC offline.
signal	Specify the type of signaling for a voice port.
snmp enable peer-trap poor-qov	Generate poor-quality-of-voice notification for applicable calls associated with VoIP dial peers.
snmp-server enable traps	Enable the router to send SNMP traps.
snmp trap link-status	Enable SNMP trap messages to be generated when this voice port is brought up or down.
timeouts initial	Configure the initial digit timeout value for a specified voice port.
timeouts interdigit	Configure the interdigit timeout value for a specified voice port.
timing	Specify timing parameters for a specified voice port.
type	Specify the E&M interface type.
vad	Enable VAD for the calls using this dial peer.
voice-port	Enter the voice port configuration mode.

A subset of the commands listed are voice-port commands. Different voice signaling types support different voice-port commands. Table 4-2 lists the router voice-port commands and the signaling types supported.

Table 4-2 Router Voice-Port Commands and Signaling Types Supported

Voice-Port Command	FXO	FXS	E&M
comfort-noise			
connection			
cptone	X	X	X
description	X	X	X
dial-type	X		X
echo-cancel coverage			
echo-cancel enable			
impedance	X	X	X
input gain	X	X	X
music-threshold			
non-linear			
operation			X
output attenuation	X	X	X
ring frequency		X	
ring number	X		
shutdown	X	X	X
signal	X	X	X
snmp trap link-status			
timeouts initial			
timeouts interdigit			
timing timing keywords:			
clear-wait			X
delay-duration			X
delay-start			X
delay-with-integrity			X
digit	X	X	X
inter-digit	X	X	X
pulse	X		X
pulse-inter-digit	X		X
wink-duration			X
wink-wait			X
type			X

Command Syntax Conventions

Table 4-3 describes the syntax used with the commands in this chapter.

Table 4-3 Command Syntax Guide

Convention	Description
boldface	Commands and keywords.
italic	Command input that is supplied by you.
[ ]	Keywords or arguments within square brackets are optional.
{ x | x | x }	A choice of keywords (represented by x) are in braces separated by vertical bars. You must select one.
^ or Ctrl	Represent the key labeled *Control*. For example, when you read *^D* or *Ctrl-D*, you should hold down the Control key while you press the D key.
`screen font`	Examples of information displayed on the screen.
`boldface screen font`	Examples of information that you must enter.
< >	Nonprinting characters, such as passwords, are in angled brackets.
[ ]	Default responses to system prompts are in square brackets.

Note Means *reader take note*. Notes contain helpful suggestions or references to additional information and material.

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

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

acc-qos

To generate an SNMP event if the QoS for a dial peer drops below a specified level, use the **acc-qos** dial-peer configuration command. Use the **no** form of this command to use the default value for this feature.

acc-qos {**best-effort** | **controlled-load** | **guaranteed-delay**}
no acc-qos

Syntax Description

best-effort	RSVP makes no bandwidth reservation.
controlled-load	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	RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.

Default

best-effort. Using the **no** form of this command is the same as the default.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **acc-qos** dial-peer command to generate an SNMP event if the QoS 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 QoS can be provided. Cisco IOS software uses RSVP to request QoS 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 QoS is not available.

This command only applies to VoIP peers.

Example

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

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

Related Commands

req-qos

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** dial-peer configuration command. Use the **no** form of this command to disable this feature.

answer-address [+]*string*
no answer-address

Syntax Description

string	Series of digits that specify the E.164 or private dialing plan telephone number: • Digits 0 through 9, letters A through D, pound sign (#), and asterisk (*), which represent specific digits that can be entered. • Plus sign (+), which is optionally used as the first digit to indicate an E.164 standard number. • Comma (,), which inserts a pause between digits. • Period (.), which is used as a wild-card character and matches any entered digit.

Default

Enabled with a null string.

Command Mode

Dial-peer configuration.

Usage Guidelines

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 dial peer associated with the interface is 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.

This command applies to both VoIP and POTS dial peers.

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.

Example

The following example configures the E.164 telephone number, *14085559626*, as the dial peer of an incoming call:

```
dial-peer voice 10 pots
 answer-address 14085559626
```

Related Commands

destination-pattern
port
prefix

codec

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

codec {**g711alaw** | **g711ulaw** | **g729r8** | **g729r8 pre-ietf**}
no codec

Syntax Description

g711alaw	G.711 A-Law 64,000 bits per second (bps).
g711ulaw	G.711 U-Law 64,000 bps.
g729r8	G.729 8000 bps.
g729r8 pre-ietf	G.729 8000 bps. Use this argument if a dial peer is using an image from Release 12.0(5)T or later.

Default

g729r8.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **codec** command to define a specific voice coder rate of speech for a dial peer.

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

If **codec**-command values for the VoIP peers of a connection do not match, the call fails.

This command only applies to VoIP peers.

Note Prior to Cisco IOS Release 12.0(5)T, **g729r8** is implemented in the pre-IETF format, thereafter it is implemented in the standard IETF format. Whenever new images, from Release 12.0(5)T or later, interoperate with older versions of VoIP (when the **g729r8** codec was not compliant with the IETF standard), users can hear garbled voices and ringback on either end of the connection. To avoid this problem, configure the dial peers with the **g729r8 pre-ietf** argument.

Example

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

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

comfort-noise

To specify whether or not background noise should be generated, use the **comfort-noise** voice-port configuration command. Use the **no** form of this command to disable this feature.

comfort-noise
no comfort-noise

Syntax Description

This command has no arguments or keywords.

Default

Enabled.

Command Mode

Voice-port configuration.

Usage Guidelines

Use the **comfort-noise** command to generate background noise to fill silent gaps during calls if VAD is activated. If **comfort noise** is not enabled and VAD is enabled at the remote end of the connection, the user hears dead silence when the remote party is not speaking.

The configuration of **comfort noise** 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.

Example

The following example enables background noise:

```
voice port 0/0
 comfort-noise
```

Related Commands

vad

connection

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

connection {**plar** | **trunk** } *string*
no connection {**plar** | **trunk** } *string*

Syntax Description

plar	Private line auto ringdown (PLAR) connection. PLAR connection associates a dial peer directly with an interface; when an interface goes off-hook, the dial peer sets up the second call leg and creates a conference call without the caller having to dial any digits.
trunk	Straight tie-line connection to a private branch exchange (PBX).
string	Destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.

Default

No connection.

Command Mode

Voice-port configuration.

Usage Guidelines

Use the **connection** command to specify a connection mode for a specific interface. 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 voice port. The destination dial peer is determined on the basis of this called number.

Use the **connection trunk** command to specify a straight tie-line connection to a PBX. This command can be used for E&M-to-E&M trunks, FXO-to-FXS trunks, and FXS-to-FXS trunks. Signaling is transported for E&M-to-E&M trunks and FXO-to-FXS trunks; signaling will not be transported for FXS-to-FXS trunks.

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

Example

The following example selects **plar** as the connection mode and a destination telephone number of *14085559262*:

```
voice port 0/0
 connection plar 14085559262
```

The following example selects **trunk** as the connection mode and a destination telephone number of *14085559262*:

```
voice port 0/0
 connection trunk 14085559262
```

Related Commands

session protocol

cptone

To configure a voice call progress tone locale, use the **cptone** voice-port configuration command. Use the **no** form of this command to disable this feature.

cptone {**australia** | **brazil** | **china** | **finland** | **france** | **germany** | **japan** | **northamerica** | **unitedkingdom**}
no cptone

Syntax Description

australia	Analog voice interface-related default tone, ring, and cadence setting for Australia.
brazil	Analog voice interface-related default tone, ring, and cadence setting for Brazil.
china	Analog voice interface-related default tone, ring, and cadence setting for China.
finland	Analog voice interface-related default tone, ring, and cadence setting for Finland.
france	Analog voice interface-related default tone, ring, and cadence setting for France.
germany	Analog voice interface-related default tone, ring, and cadence setting for Germany.
japan	Analog voice interface-related default tone, ring, and cadence setting for Japan.
northamerica	Analog voice interface-related default tone, ring, and cadence setting for North America.
unitedkingdom	Analog voice interface-related default tone, ring, and cadence setting for the United Kingdom.

Default

northamerica.

Command Mode

Voice-port configuration.

Usage Guidelines

Use the **cptone** command to specify a regional analog voice interface-related 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.

Example

The following example configures North America as the call progress tone locale:

```
voice port 0/0
 cptone northamerica
```

description

To include a description of what this voice port is connected to, use the **description** voice-port configuration command. Use the **no** form of this command to disable this feature.

description *string*
no description

Syntax Description

string	Character string from 1 to 255 characters.

Default

Enabled with a null string.

Command Mode

Voice-port configuration.

Usage Guidelines

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.

Example

The following example identifies this voice port as a connection to the purchasing department:

```
voice port 0/0
 description purchasing_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** dial-peer configuration command. Use the **no** form of this command to disable this feature.

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

Syntax Description

string	Series of digits that specify the E.164 or private dialing plan telephone number: • Digits 0 through 9, letters A through D, pound sign (#), and asterisk (*), which represent specific digits that can be entered. • Plus sign (+), which is optionally used as the first digit to indicate an E.164 standard number. • Comma (,), which inserts a pause between digits. • Period (.), which is used as a wild-card character and matches any entered digit.

Default

Enabled with a null string.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **destination-pattern** command to define the E.164 telephone number for this dial peer. This pattern is used to match dialed digits to a dial peer. The dial peer is then used to complete the call.

This command applies to both VoIP and POTS dial peers.

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.

Example

The following example configures the E.164 telephone number, *14085557922*, for a dial peer:

```
dial-peer voice 10 pots
 destination-pattern 14085557922
```

Related Commands

answer-address
prefix

dial-control-mib

To specify attributes for the call history table, use the **dial-control-mib** global configuration command.

dial-control-mib {**max-size** *number* | **retain-timer** *number*}

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Syntax Description

max-size *number*	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*	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 Mode

Global configuration.

Usage Guidelines

The call history table contains a listing of all calls connected through the router in descending time order since VoIP was enabled. Use the **dial-control-mib** global configuration command to specify attributes for the call history table.

Example

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 voice

To enter the dial peer configuration mode (and specify the method of voice-related encapsulation), use the **dial-peer voice** global configuration command.

dial-peer voice *number* {**voip** | **pots**}

Syntax Description

number	Digit(s) defining a particular dial peer. Valid entries are from 1 to 2147483647.
voip	VoIP dial peer using voice encapsulation on the POTS network.
pots	POTS dial peer using VoIP encapsulation on the IP backbone.

Default

No dial peer configuration mode is preconfigured.

Command Mode

Global configuration.

Usage Guidelines

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

Example

The following example accesses the dial peer configuration mode and configures a POTS dial peer identified as dial peer 10:

```
configure terminal
 dial-peer voice 10 pots
```

Related Commands

voice-port

dial-type

To specify the type of out-dialing for voice-port interfaces, use the **dial-type** voice-port configuration command. Use the **no** form of this command to disable this feature.

dial-type {**dtmf** | **pulse**}
no dial-type

Syntax Description

dtmf	Touch-tone dialer.
pulse	Pulse dialer.

Default

dtmf.

Command Mode

Voice-port configuration.

Usage Guidelines

Use the **dial-type** command to specify an out-dialing type for an FXO or E&M voice-port interface; this command does not apply 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.

Example

The following example configures a voice port to support a touch-tone dialer:

```
voice port 0/0
 dial-type dtmf
```

echo-cancel coverage

To adjust the size of the echo cancel, use the **echo-cancel coverage** voice-port configuration command. Use the **no** form of this command to reset this command to the default value.

echo-cancel coverage *value*
no echo-cancel coverage *value*

Syntax Description

value	Number of milliseconds (ms) the echo-canceller covers on a given signal. Valid values are 8, 16, 24, and 32 ms.

Default

16 ms.

Command Mode

Voice-port configuration.

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 of the interface and received back on the same interface within the configured amount of time. If the local loop (the distance from the analog 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, the echo canceller takes longer to converge; in this case, the user might hear a 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 IP side of the connection.

Note This command is valid only if the echo cancel feature has been enabled. For more information, refer to the **echo-cancel enable** command.

Example

The following example adjusts the size of the echo canceller to 16 ms:

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

Related Commands

echo-cancel enable

echo-cancel enable

To enable the echo cancel feature, use the **echo-cancel enable** voice-port configuration command. Use the **no** form of this command to disable this feature.

echo-cancel enable
no echo-cancel enable

Syntax Description

This command has no arguments or keywords.

Default

Enabled for all interface types.

Command Mode

Voice-port configuration.

Usage Guidelines

The **echo-cancel** command enables cancellation of voice that is sent out of the interface and is received back on the same interface. 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** command does not affect the echo heard by the user on the analog side of the connection.

There is no echo path for a four-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.

Example

The following example enables the echo cancel feature for 16-millisecond echo coverage:

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

Related Commands

echo-cancel coverage
non-linear

expect-factor

To specify when the router generates an alarm to the network manager, indicating that the expected quality of voice has dropped, use the **expect-factor** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

expect-factor *value*
no expect-factor *value*

Syntax Description

value	Integers that represent the ITU-T specification for quality of voice as described in G.113. Valid entries are from 0 to 20, with 0 representing toll quality.

Default

10.

Command Mode

Dial-peer configuration.

Usage Guidelines

VoIP monitors the quality of voice received over the network. Use the **expect-factor** command to specify when the router generates an SNMP trap to the network manager.

This command only applies to VoIP peers.

Example

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 fascimile (fax) is sent to the specified dial peer, use the **fax-rate** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

fax-rate{**2400** | **4800** | **7200** | **9600** | **14400** | **disable** | **voice**}
no fax-rate

Syntax Description

2400	Fax transmission speed of 2400 bps.
4800	Fax transmission speed of 4800 bps.
7200	Fax transmission speed of 7200 bps.
9600	Fax transmission speed of 9600 bps.
14400	Fax transmission speed of 14,400 bps.
disable	Fax relay transmission capability disabled.
voice	Highest possible transmission speed allowed by voice rate.

Default

voice.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **fax-rate** command to specify the fax transmission rate to the specified dial peer.

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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** command is set above the **codec** command rate in the same dial peer, the data sent over the network for fax transmission exceeds the bandwidth reserved for RVSP. Because more network bandwidth is monopolized by the fax transmission, we do not recommend setting the **fax-rate** value higher than the **codec** command value. If the **fax-rate** value is set lower than the **codec**-command value, faxes take longer to transmit but use less bandwidth.

This command only applies to VoIP peers.

Example

The following example configures a fax rate of 9600 bps for faxes sent to a dial peer:

```
dial-peer voice 10 voip
 fax-rate 9600
```

Related Commands

codec

icpif

To specify the Calculated Planning Impairment Factor (ICPIF) for calls sent by a dial peer, use the **icpif** dial-peer configuration command. Use the **no** form of this command to restore the default value for this command.

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.

Default

30 equipment impairment factor units.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **icpif** command to specify the maximum acceptable impairment factor for the voice calls sent by the selected dial peer.

This command only applies to VoIP peers.

Example

The following example disables the **icpif** command:

```
dial-peer voice 10 voip
 icpif 0
```

impedance

To specify the terminating impedance of a voice-port interface, use the **impedance** voice-port configuration command. Use the **no** form of this command to restore the default value.

impedance {**600c** | **600r** | **900c** | **complex1** | **complex2**}
no impedance

Syntax Description

600c	600 ohms complex.
600r	600 ohms real.
900c	900 ohms complex.
complex1	Complex 1.
complex2	Complex 2.

Default

600 ohms.

Command Mode

Voice-port configuration.

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), a significant amount of echo is 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 changes the impedance on both voice ports of a VIC. This voice port must be shut down and then opened for the new value to take effect.

This command applies to FXS, FXO, and E&M voice ports.

Example

The following example configures an FXO voice port for a terminating impedance of 600 ohms:

```
voice port 0/0
impedance 600r
```

input gain

To configure a specific input gain value, use the **input gain** voice-port configuration command. Use the **no** form of this command to disable this feature.

input gain *value*

no input gain *value*

Syntax Description

value	Amount of gain in decibels (dB) to be inserted at the receiver side of the interface. Acceptable value is any integer from –6 to 14.

Default

0 dB.

Command Mode

Voice-port configuration.

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. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that, normally, there must be –6 dB of 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.

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 into the router. If the voice level is too low, you can increase the input gain.

Example

The following example configures a 3-dB gain for the receiver side of the interface:

```
voice port 0/0
input gain 3
```

Related Commands

output attenuation

ip precedence

To set IP precedence (priority) for packets sent by the dial peer, use the **ip precedence** dial-peer configuration command. Use the **no** form of this command to restore the default value for this command.

ip precedence *number*
no ip precedence

Syntax Description

number	Integer specifying the IP precedence value. Valid entries are 0 to 7. A value of 0 means that no precedence (priority) has been set.

Default

No precedence (0).

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **ip precedence** 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 QoS for voice packets need to have a higher priority than other IP packets. The **ip precedence** 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 only applies to VoIP peers.

Example

The following example sets the IP precedence at 5:

```
dial-peer voice 10 voip
 ip precedence 5
```

ip udp checksum

To calculate the User Datagram Protocol (UDP) checksum for voice packets transmitted by the dial peer, use the **ip udp checksum** dial-peer configuration command. 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.

Default

Disabled.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **ip udp checksum** command to enable UDP checksum calculation for each outbound voice packet. 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 bad voice packets forwarded to the DSP.

This command only applies to VoIP peers.

Example

The following example calculates the UDP checksum for voice packets transmitted by this dial peer:

```
dial-peer voice 10 voip
 ip udp checksum
```

music-threshold

To specify the threshold for on-hold music for a specified voice port, use the **music-threshold** voice-port configuration command. Use the **no** form of this command to disable this feature.

music-threshold *number*
no music-threshold *number*

Syntax Description

number	On-hold music threshold in dB. Valid entries are any integer from –70 to –30.

Default

–38 dB.

Command Mode

Voice-port configuration.

Usage Guidelines

Use the **music-threshold** command to specify the dB level of music played when calls are 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.

Example

The following sets the dB threshold for the music played when calls are put on hold to –35:

```
voice port 0/0
 music-threshold —35
```

non-linear

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

non-linear
no non-linear

Syntax Description

This command has no arguments or keywords.

Default

Enabled.

Command Mode

Voice-port configuration.

Usage Guidelines

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 hear truncation of consonants at the end of sentences when this command is enabled.

This feature is also generally known as residual echo suppression.

Example

The following example enables nonlinear call processing:

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

Related Commands

echo-cancel enable

num-exp

To define how to expand an extension number into a particular destination pattern, use the **num-exp** global configuration command.

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.

Default

No number expansions are predefined.

Command Mode

Global configuration.

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 by using variables. You can also use this command to convert seven-digit numbers to numbers of less than seven digits.

Use a period (.) as a variable or wildcard 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, enter four periods.

Examples

The following example expands the extension number *54001* to *14085554001*:

```
num-exp 54001 14085554001
```

The following example shows how to expand all five-digit extensions beginning with 5 and append the extension numbers to *1408555*:

```
num-exp 5.... 1408555....
```

operation

To select a specific cabling scheme for E&M ports, use the **operation** voice-port configuration command. Use the **no** form of this command as an alternative method of configuring two-wire operation.

operation {**2-wire** | **4-wire**}
no operation {**2-wire** | **4-wire**}

Syntax Description

2-wire	Two-wire E&M cabling scheme.
4-wire	Four-wire E&M cabling scheme.

Default

2-wire.

Command Mode

Voice-port configuration.

Usage Guidelines

The **operation** command only affects voice traffic. Signaling is independent of two-wire versus four-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 VIC. The voice port must be shut down and then opened again for the new value to take effect.

This command does not apply to FXS or FXO interfaces because those are, by definition, two-wire interfaces.

Example

The following example specifies that an E&M port uses a four-wire cabling scheme:

```
voice port 0/0
 operation 4-wire
```

output attenuation

To configure a specific output attenuation value, use the **output attenuation** voice-port configuration command. Use the **no** form of this command to disable this feature.

output attenuation *value*
no output attenuation

Syntax Description

value	Amount of attenuation in dB at the transmit side of the interface. Acceptable value is any integer from 0 to 14.

Default

0 dB.

Command Mode

Voice-port configuration.

Usage Guidelines

A system-wide loss plan must be implemented by 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. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that, normally, there must be –6 dB of 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.

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.

Example

The following example configures a 3-dB gain to be inserted at the transmit side of the interface:

```
voice port 0/0
 output attenuation 3
```

Related Commands

input gain

port

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

port *slot-number/port*
no port

Syntax Description

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	Voice port. Valid entries are 0 or 1.

Default

No port is preconfigured.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **port** configuration command to associate the designated voice port with the selected dial peer.

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 only applies to POTS peers.

Example

The following example associates a dial peer with slot 0 and access through port 0:

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

prefix

To specify the prefix of the dialed digits for this dial peer, use the **prefix** dial-peer configuration command. 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.

Default

Null string.

Command Mode

Dial-peer configuration.

Usage Guidelines

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 first sent to the telephony interface, before the telephone number is 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.

This command only applies to POTS peers.

Example

The following example specifies a prefix of *9* and then a pause:

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

Related Commands

answer-address
destination-pattern

req-qos

To specify the desired QoS to be used in reaching a specified dial peer, use the **req-qos** dial-peer configuration command. Use the **no** form of this command to restore the default value for this command.

req-qos {**best-effort** | **controlled-load** | **guaranteed-delay**}
no req-qos

Syntax Description

best-effort	RSVP makes no bandwidth reservation.
controlled-load	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	RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.

Default

best-effort. The **no** form of this command restores the default value.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **req-qos** command to request a specific QoS to be used in reaching a dial peer. This command is like **acc-qos**; the software reserves a certain amount of bandwidth to provide the selected QoS. Cisco IOS software uses RSVP to request QoS guarantees from the network.

This command only applies to VoIP peers.

Example

The following example configures guaranteed-delay as the desired (requested) QoS to a dial peer:

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

Related Commands

acc-qos

ring frequency

To specify the ring frequency for a specified FXS voice port, use the **ring frequency** voice-port configuration command. Use the **no** form of this command to reset the default value for this command.

ring frequency *number*
no ring frequency

Syntax Description

number	Ring frequency in Hz used in the FXS interface. Valid entries are 25 and 50 Hz.

Default

25 Hz.

Command Mode

Voice-port configuration.

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. 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.

Example

The following example configures the ring frequency for 50 Hz:

```
voice port 0/0
 ring frequency 50
```

Related Commands

ring number

ring number

To specify the number of rings for a specified FXO voice port, use the **ring number** voice-port configuration command. Use the **no** form of this command to reset the default value for 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.

Default

1 ring.

Command Mode

Voice-port configuration.

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.

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 other equipment on line did not answer the incoming call in the configured number of rings.

This command does not apply to FXS or E&M interfaces because they do not receive ringing to receive a call.

Example

The following example sets five rings as the maximum number of rings to be detected before closing a connection over this voice port:

```
voice port 0/0
 ring number 5
```

Related Commands

ring frequency

session protocol

To establish a session protocol for calls between the local and remote routers via the packet network, use the **session protocol** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

session protocol cisco
no session protocol

Syntax Description

cisco	Cisco Session Protocol.

Default

cisco.

Command Mode

Dial-peer configuration.

Usage Guidelines

For this release, **cisco** is the only applicable session protocol. This command only applies to VoIP peers.

Example

The following example selects Cisco Session Protocol as the session protocol:

```
dial-peer voice 10 voip
 session protocol cisco
```

Related Commands

session target

session target

To specify a network-specific address for a specified dial peer, use the **session target** dial-peer configuration command. Use the **no** form of this command to disable this feature.

session target {**ipv4:***destination-address* | **dns:**[**s.** | **d.** | **u.**] *host-name* | **loopback:rtp** | **loopback:compressed** | **loopback:uncompressed**}
no session target

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Syntax Description

ipv4:*destination-address*	IP address of the dial peer.
dns:host-name	Domain name system (DNS) server is 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 wildcards with this keyword when defining the session target for VoIP dial peers: • **s.**—Source destination pattern is used as part of the domain name. • **d.**—Destination number is used as part of the domain name. • **u.**—Unmatched portion of the destination pattern (such as a defined extension number) is used as part of the domain name.
loopback:rtp	All voice data is looped-back to the originating source. This only applies to VoIP dial peers.
loopback:compressed	All voice data is looped-back in compressed mode to the originating source. This only applies to POTS dial peers.
loopback:uncompressed	All voice data is looped-back in uncompressed mode to the originating source. This only applies to POTS dial peers.

Default

Enabled with no IP address or domain name defined.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **session target** command to specify a network-specific address or domain name for a dial peer.

The **session target loopback** command is used for testing the voice transmission path of a call. The loopback point depends on the call origination and the loopback type selected.

The **session target dns** command can be used with or without the specified wildcards. The optional wildcards reduce the number of VoIP dial-peer session targets you need to configure if you have groups of numbers associated with a particular router.

Example

The following example configures a session target using **dns** for hostname *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** and 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.** means that the router uses 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.** means that the router uses 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
```

Related Commands

destination-pattern
session protocol

show call active voice

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

show call active voice

Syntax Description

This command contains no arguments or keywords.

Command Mode

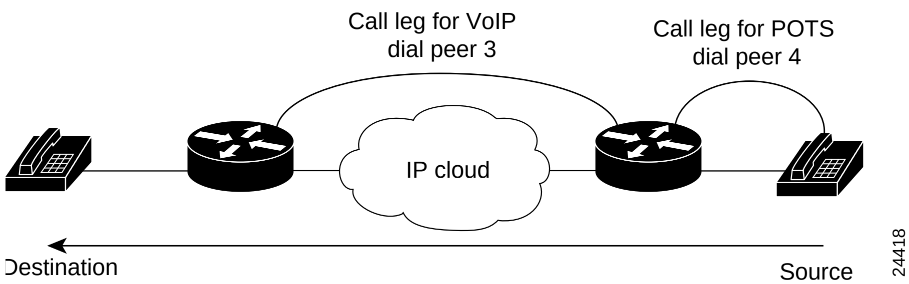
Privileged EXEC.

Usage Guidelines

Use the **show call active voice** privileged EXEC command to display the contents of the active call table, which shows all of the calls currently connected through the router.

For each call, there are two call legs, a POTS call leg and a VoIP call leg. A call leg is a discrete segment of a call between two points in the connection. Each dial peer creates a call leg, as shown in Figure 4-1.

Figure 4-1 Call Legs Example



These two call legs are associated by the connection ID. The connection ID is global across the voice network so that you can associate two call legs on one router with two call legs on another router, thereby providing an end-to-end view of a call.

Sample Display

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

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

VOIP: ConnectionId[0x19BDF910 0xAF500007 0x0 0x58ED0] RemoteIPAddress=17635075
RemoteUDPPort=16394 RoundTripDelay=0 SelectedQoS=0 SessionProtocol=1
SessionTarget= OnTimeRvPlayout=0 GapFillWithSilence=0 GapFillWithPrediction=600
GapFillWithInterpolation=0 GapFillWithRedundancy=0 HiWaterPlayoutDelay=110
LoWaterPlayoutDelay=64 ReceiveDelay=94 VADEnable=0 CoderTypeRate=0

GENERIC: SetupTime=21072 Index=1 PeerAddress=14085554001 PeerSubAddress=
PeerId=0 PeerIfIndex=0 LogicalIfIndex=5 ConnectTime=21115 CallState=4 CallOrigin=1
ChargedUnits=0 InfoType=1 TransmitPackets=377915 TransmitBytes=7558300
ReceivePackets=375594 ReceiveBytes=7511880

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

Table 4-4 provides an alphabetical listing of the fields in this output and a description of each field.

Table 4-4 Show-Call-Active-Voice Command Field Descriptions

Field	Description
ACOM Level	Current ACOM level for the call. This value is sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CallState	Current state of the call.
CoderTypeRate	Negotiated coder transmit rate of voice/fax compression during the call.
ConnectionId	Global call identifier of a gateway call.
ConnectTime	Time at which the call was connected.

Table 4-4 Show-Call-Active-Voice Command Field Descriptions (Continued)

Field	Description
Dial-Peer	Tag of the dial peer transmitting this call.
ERLLevel	Current Echo Return Loss (ERL) level for this call.
FaxTxDuration	Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWith Silence	Duration of voice signal replaced with silence because voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call. An example of such pullout is frame-eraser or frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received on time from voice gateway for this call.
GapFillWith Redundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because voice data was lost or not received on time from voice gateway for this call.
HiWaterPlayoutDelay	High-water mark Voice Playout FIFO Delay during this call.
Index	Dial-peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call.
InSignalLevel	Active input signal level from the telephony interface used by this call.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low-water mark Voice Playout FIFO Delay during the call.
NoiseLevel	Active noise level for the call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
OutSignalLevel	Active output signal level to telephony interface used by this call.
PeerAddress	Destination pattern associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice-port index number for this peer.
PeerSubaddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the decoder delay during the voice call.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system UDP listener port to which voice packets are transmitted.

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Table 4-4 Show-Call-Active-Voice Command Field Descriptions (Continued)

Field	Description
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone during the call.
SelectedQoS	Selected RSVP QoS for the call.
SessionProtocol	Session protocol used for an Internet call between the local and remote router via the IP backbone.
SessionTarget	Session target of the peer used for the call.
SetupTime	Value of the System UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted from this peer during the call.
TransmitPackets	Number of packets transmitted from this peer during the call.
TxDuration	Duration of transmit path open from this peer to the voice gateway for the call.
VADEnable	Whether or not 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

show call history voice
show dial-peer voice
show num-exp
show voice port

show call history voice

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

show call history voice last *number*

Syntax Description

last *number*	Displays the last calls connected, where the number of calls displayed is defined by the argument *number.* Valid entries for the argument *number* is any number from 1 to 2147483647.

Command Mode

Privileged EXEC.

Usage Guidelines

Use the **show call history voice** privileged EXEC command to display the call history table. The call history table contains a listing of all calls connected through this router in descending time order since VoIP 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.*

Sample Display

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

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

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

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

Table 4-5 provides an alphabetical listing of the fields in this output and a description of each field.

Table 4-5 Show-Call-History-Voice Command Field Descriptions

Field	Description
ACOMLevel	Average ACOM level for this call. This value is sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CoderTypeRate	Negotiated coder rate. This value specifies the transmit rate of voice/fax compression to its associated call leg for the call.
ConnectionID	Global call identifier for the gateway call.
ConnectTime	Time the call was connected.
DisconnectCause	Description explaining why the call was disconnected.
DisconnectText	Descriptive text explaining the disconnect reason.
DisconnectTime	Time the call was disconnected.
FaxDuration	Duration of fax transmitted from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing this value by the TxDuration value.
GapFillWithSilence	Duration of voice signal replaced with silence because the voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.

Table 4-5 Show-Call-History-Voice Command Field Descriptions (Continued)

Field	Description
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.
GapFillWithRedundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because the voice data was lost or not received on time from the voice gateway for this call.
HiWaterPlayoutDelay	High-water mark Voice Playout FIFO Delay during the voice call.
Index	Index number identifying the voice-peer for this call.
InfoType	Information type for this call.
LogicalIfIndex	Index of the logical voice port for this call.
LoWaterPlayoutDelay	Low-water mark Voice Playout FIFO Delay during the voice call.
NoiseLevel	Average noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
PeerAddress	Destination pattern or number to which this call is connected.
PeerId	ID value of the peer entry table to which this call was made.
PeerIfIndex	Index number of the logical interface through which this call was made. For ISDN media, this would be the index number of the B channel used for the call.
PeerSubAddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the decoder delay during the voice call.
ReceivePackets	Number of packets received by this peer during the call.
RemoteIPAddress	Remote system IP address for the call.
RemoteUDPPort	Remote system UDP listener port to which voice packets for this call are transmitted.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone for this call.
SelectedQoS	Selected RSVP QoS for the 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 the call.
SetUpTime	Value of the System UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted by this peer during the call.
TransmitPackets	Number of packets transmitted by this peer during the call.
TxDuration	Duration of the transmit path open from this peer to the voice gateway for the call.

Table 4-5 Show-Call-History-Voice Command Field Descriptions (Continued)

Field	Description
VADEnable	Whether or not VAD was enabled for this call.
VoiceTxDuration	Duration of voice transmitted from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration by the TxDuration value.

Related Commands

show call active voice
show dial-peer voice
show num-exp
show voice port

show controllers voice

To display information about voice related hardware, use the **show controllers voice** privileged EXEC command.

show controllers voice

Syntax Description

This command contains no arguments or keywords.

Command Mode

Privileged EXEC.

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 DSP, and the DSP firmware versions.

The information displayed is generally useful for diagnostic tasks performed by technical support people only.

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Sample Display

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

```
router# show controllers voice
EPIC Switch registers:
STDA 0xFF STDB 0x0 SARA 0x0 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 0x1 pRemoveRx 0x1 pInsertTx 0x2 pRemoveTx 0x2

Rx Message 0:
packet_length 12 channel_id 0 packet_id 6 process id1 0xFECE process id2 0xFACE
0000:0000

Rx Message 1:
packet_length 12 channel_id 0 packet_id 6 process id1 0xFECE process id2 0xFACE
0000:0000

Rx Message 2:
packet_length 12 channel_id 0 packet_id 6 process id1 0xFECE process id2 0xFACE
0000:0000

 --More--         Rx Message 3:
packet_length 12 channel_id 0 packet_id 6 process id1 0xFECE process id2 0xFACE
0000:0000

Rx Message 4:
packet_length 12 channel_id 0 packet_id 6 process id1 0xFECE process id2 0xFACE
0000:0000

Rx Message 5:
packet_length 12 channel_id 0 packet_id 6 process id1 0xFECE process id2 0xFACE
0000:0000

Tx Message 0:
packet_length 66 channel_id 0 packet_id 198 process id1 0xFECE process id2 0xFACE
0000:0000 0000 0000 0000 0042 003F 0000 0000 0000 0000
0020:0000 0006 0006 0006 0006 0006 0006 0006 0006 0006
0040:0006 0006 0006 0006 0006 0006 0006 0000

Tx Message 1:
packet_length 66 channel_id 0 packet_id 198 process id1 0xFECE process id2 0xFACE
0000:0000 0000 0000 0000 0043 0040 0000 0000 0000 0000
 --More--         0020:0000 0006 0006 0006 0006 0006 0006 0006 0006 0006
0040:0006 0006 0006 0006 0006 0006 0006 0000

Tx Message 2:
packet_length 66 channel_id 0 packet_id 198 process id1 0xFECE process id2 0xFACE
0000:0000 0000 0000 0000 003B 0038 0000 0000 0000 0000
0020:0000 0006 0006 0006 0006 0006 0006 0006 0006 0006
0040:0006 0006 0006 0006 0006 0006 0006 0000

Tx Message 3:
packet_length 66 channel_id 0 packet_id 198 process id1 0xFECE process id2 0xFACE
0000:0000 0000 0000 0000 003C 0039 0000 0000 0000 0000
0020:0000 0006 0006 0006 0006 0006 0006 0006 0006 0006
0040:0006 0006 0006 0006 0006 0006 0006 0000
```

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```
Tx Message 4:
packet_length 66 channel_id 0 packet_id 198 process id1 0xFECE process id2 0xFACE
0000:0000 0000 0000 0000 003D 003A 0000 0000 0000 0000
0020:0000 0006 0006 0006 0006 0006 0006 0006 0006 0006
0040:0006 0006 0006 0006 0006 0006 0006 0000
 --More--
Tx Message 5:
packet_length 66 channel_id 0 packet_id 198 process id1 0xFECE process id2 0xFACE
0000:0000 0000 0000 0000 003E 003B 0000 0000 0000 0000
0020:0000 0006 0006 0006 0006 0006 0006 0006 0006 0006
0040:0006 0006 0006 0006 0006 0006 0006 0000

Tx Message 6:
packet_length 66 channel_id 0 packet_id 198 process id1 0xFECE process id2 0xFACE
0000:0000 0000 0000 0000 003F 003C 0000 0000 0000 0000
0020:0000 0006 0006 0006 0006 0006 0006 0006 0006 0006
0040:0006 0006 0006 0006 0006 0006 0006 0000

Tx Message 7:
packet_length 66 channel_id 0 packet_id 198 process id1 0xFECE process id2 0xFACE
0000:0000 0000 0000 0000 0040 003D 0000 0000 0000 0000
0020:0000 0006 0006 0006 0006 0006 0006 0006 0006 0006
0040:0006 0006 0006 0006 0006 0006 0006 0000

Tx Message 8:
 --More--         packet_length 66 channel_id 0 packet_id 198 process id1 0xFECE process
id2 0xFACE
0000:0000 0000 0000 0000 0041 003E 0000 0000 0000 0000
0020:0000 0006 0006 0006 0006 0006 0006 0006 0006 0006
0040:0006 0006 0006 0006 0006 0006 0006 0000

Bootloader 1.8, Appn 3.1
Application firmware 3.1.1, Built by claux on Mon Mar 22 16:32:13 1999

VIC Interface Foreign Exchange Station 1/0, DSP instance (0x19355C0)
 Singalling channel num 128 Signalling proxy 0x0 Signaling dsp 0x19355C0
 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
 264434 bytes output, 1036 frames output, 0 output errors, 0 output underrun
 0 unaligned frames

VIC Interface Foreign Exchange Station 1/1, DSP instance (0x19357F0)
 Singalling channel num 129 Signalling proxy 0x0 Signaling dsp 0x19357F0
 tx outstanding 0, max tx outstanding 32
 ptr 0x0, length 0x0, max length 0x0
 --More--         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
 68 bytes output, 4 frames output, 0 output errors, 0 output underrun
 0 unaligned frames
```

show diag

To display hardware information for the router, use the **show diag** privileged EXEC command.

show diag

Syntax Description

This command contains no arguments or keywords.

Command Mode

Privileged EXEC.

Usage Guidelines

This command displays information for the electrically erasable programmable read-only memory (EEPROM), motherboard, and the WAN interface cards and voice interface cards (WICs/VICs).

Sample Display

The following is sample output from the **show diag** command:

```
router# show diag
Slot 0:
        C1750 1FE VE Mainboard port adapter, 6 ports
        Port adapter is analyzed
        Port adapter insertion time unknown
        EEPROM contents at hardware discovery:
        Hardware revision 0.0           Board revision UNKNOWN
        Serial number     1314672220    Part number    00-0000-00
        Test history      0x0           RMA number     00-00-00
        EEPROM format version 1
        EEPROM contents (hex):
          0x20:01 C9 00 00 4E 5C 4E 5C 00 00 00 00 00 00 00 00
          0x30:00 00 00 04 00 00 00 00 00 00 00 00 00 00 00 00

Packet Voice DSP Module:
        Hardware Revision        :1.0
        Board Revision           :01
        Processor type           :02
        Part Number              :73-3933-01
        Number of DSP's          :2
        Type of DSP              :TMS320C549
        EEPROM format version 4
        EEPROM contents (hex):
        0x00:  04 FF 40 01 5B 41 01 00 42 30 31 09 02 82 49 0F
        0x10:  5D 01 FF

        WIC Slot 0:
        BRI U - 2091 WAN daughter card
        Hardware revision 1.3    Board revision A0
        Serial number     0004147773    Part number    800-01834-01
        Test history      0x00          RMA number     00-00-00
        Connector type    WAN Module
        EEPROM format version 1
        EEPROM contents (hex):
        0x20:  01 09 01 03 00 3F 4A 3D 50 07 2A 01 00 00 00 00
        0x30:  50 00 00 00 96 11 06 01 FF FF FF FF FF FF FF FF

        WIC Slot 1:
        Dual FXS Voice Interface Card WAN daughter card
        Hardware revision 1.1    Board revision C0
        Serial number     0010377882    Part number    800-02493-01
        Test history      0x00          RMA number     00-00-00
        Connector type    WAN Module
        EEPROM format version 1
        EEPROM contents (hex):
        0x20:  01 0E 01 01 00 9E 5A 9A 50 09 BD 01 00 00 00 00
        0x30:  60 00 00 00 98 09 10 01 FF FF FF FF FF FF FF FF

        WIC Slot 2:
        Dual EAM Voice Interface Card WAN daughter card
        Hardware revision 1.1    Board revision C0
        Serial number     0009886880    Part number    800-02497-01
        Test history      0x00          RMA number     00-00-00
        Connector type    WAN Module
        EEPROM format version 1
        EEPROM contents (hex):
        0x20:  01 0F 01 01 00 96 DC A0 50 09 C1 01 00 00 00 00
        0x30:  60 00 00 00 98 08 26 01 FF FF FF FF FF FF FF FF
Message-ID:<37014A10.3506648@cisco.com>
```

show dial-peer voice

To display configuration information for dial peers, use the **show dial-peer voice** privileged EXEC command.

show dial-peer voice [*number*]

Syntax Description

number	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.

Command Mode

Privileged EXEC.

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.

Sample Display

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 = `14085551000',
        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 = `14085',
        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 4-6 explains the fields contained in both of these examples.

Table 4-6 Show-Dial-Peer-Voice Command Field Descriptions

Field	Description
AcceptedCalls	Number of calls from this peer accepted since system startup.
acc-qos	Lowest acceptable QoS 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.
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.
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 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 is 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.
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.

Table 4-6 Show-Dial-Peer-Voice Command Field Descriptions (Continued)

Field	Description
req-qos	Configured requested QoS 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 VAD is enabled for this dial peer.

Related Commands

show call active voice
show call-history voice
show num-exp
show voice port

show dialplan incall number

To pair different voice ports and telephone numbers together for troubleshooting, use the **show dialplan incall number** privileged EXEC command.

show dialplan incall *slot-number/port* **number** *dial string*

Syntax Description

slot-number	Slot number in the router where the VIC is installed. Valid entries are from 0 to 2, depending on the VIC you have installed.
port	Voice port. Valid entries are 0 or 1.
dial string	Particular destination pattern (telephone number).

Command Mode

Privileged EXEC.

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.

Example

The following example tests whether the telephone extension *57681* can be reached through voice port 0/1:

```
show dialplan incall 0/1 number 57681
```

Related Commands

show dialplan number

show dialplan number

To show which dial peer is reached when a particular telephone number is dialed, use the **show dial plan number** privileged EXEC command.

show dial plan number *dial string*

Syntax Description

dial string	Particular destination pattern (telephone number).

Command Mode

Privileged EXEC.

Usage Guidelines

Use the **show dialplan number** command to test that the dial-plan configuration is valid and working as expected.

Example

The following example displays the dial peer associated with the destination pattern of 54567:

```
show dialplan number 54567
```

Related Commands

show dialplan incall number

show num-exp

To show the number expansions configured, use the **show num-exp** privileged EXEC command.

show num-exp [*dialed- number*]

Syntax Description

dialed-number	Displays number expansion for the specified dialed number.

Command Mode

Privileged EXEC.

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.

Sample Display

The following is sample output from the **show num-exp** command:

```
router# show num-exp
Dest Digit Pattern = '0...'     Translation = '14085550...'
Dest Digit Pattern = '1...'     Translation = '14085551...'
Dest Digit Pattern = '3..'      Translation = '140855503..'
Dest Digit Pattern = '4..'      Translation = '140855504..'
Dest Digit Pattern = '5..'      Translation = '140855505..'
Dest Digit Pattern = '6....'    Translation = '1408526....'
Dest Digit Pattern = '7....'    Translation = '1408527....'
Dest Digit Pattern = '8...'     Translation = '14085558...'
```

Table 4-6 explains the fields in the sample output.

Table 4-7 Show-Dial-Peer-Voice Command 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

show call active voice
show call history voice
show dial-peer voice
show voice port

show voice dsp

To show the current status of all DSP voice channels, use the **show voice dsp** privileged EXEC command.

show voice dsp

Syntax Description

This command has no arguments or keywords.

Command Mode

Privileged EXEC.

Usage Guidelines

This command also applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Sample Display

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

```
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 4-8 explains the fields in the sample output.

Table 4-8 Show Voice DSP Command Field Descriptions

Field	Description
DSP	Number of the DSP.
Channel	Number of the channel and its status.

Related Commands

show dial-peer voice
show voice call summary
show voice port

show voice port

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

show voice port *slot-number/port*

Syntax Description

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	Voice port. Valid entries are 0 or 1.

Command Mode

Privileged EXEC.

Usage Guidelines

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

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Sample Display

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

```
router# show voice port 0/0
E&M Slot 0/0
 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:

```
router# show voice port 0/0
Foreign Exchange Station 0/0 Slot 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
```

Table 4-9 explains the fields in the sample output.

Table 4-9 Show-Voice-Port Command Field Descriptions

Field	Description
Administrative State	Administrative state of the voice port.
Alias	User-supplied alias for this voice port.
Clear Wait Duration Timing	Time of inactive seizure signal to declare call cleared.
Connection Mode	Connection mode of the interface.
Connection Number	Full E.164 telephone number used to establish a connection with the trunk or PLAR mode.
Currently Processing	Type of call currently being processed: none, voice, or fax.
Delay Duration Timing	Maximum delay signal duration for delay dial signaling.
Delay Start Timing	Timing of generation of delayed start signal from detection of incoming seizure.
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.

Table 4-9 Show-Voice-Port Command Field Descriptions (Continued)

Field	Description
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.
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.
Number of signaling protocol errors	Number of signaling protocol errors.
Non-Linear Processing	Whether or not nonlinear processing is enabled for this port.
Operations State	Operation state of the port.
Operation Type	Operation of the E&M signal: two-wire or four-wire.
Out Attenuation	Amount of attenuation inserted at the transmit side of the interface.
Out Seizure	Outgoing seizure state of the E&M interface.
Port	Port number for this interface associated with the VIC.
Pulse Rate Timing	Pulse dialing rate in pulses per second (pps).
Regional Tone	Configured regional tone for this interface.
Ring Active Status	Ring active indication.
Ring Frequency	Configured ring frequency for this interface.
Ring Ground Status	Ring ground indication.
Signal Type	Type of signaling for a voice port: loop-start, ground-start, wink-start, immediate, and delay-dial.
Slot	Slot used in the VIC 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.

Table 4-9 Show-Voice-Port Command Field Descriptions (Continued)

Field	Description
Wink Duration Timing	Maximum wink duration for wink start signaling.
Wink Wait Duration Timing	Maximum wink wait duration for wink start signaling.

Related Commands

show call active voice
show call history voice
show dial-peer voice
show num-exp

shutdown (dial-peer configuration)

To change the administrative state of the selected dial peer from up to down, use the **shutdown** dial-peer configuration command. 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.

Default

No state is predefined.

Command Mode

Dial-peer configuration.

Usage Guidelines

When a dial peer is shut down, you cannot initiate calls to that peer. This command applies to both VoIP and POTS peers.

Example

The following example changes the administrative state of voice telephony dial peer 10 to down:

```
configure terminal
 dial-peer voice 10 pots
 shutdown
```

shutdown (voice-port configuration)

To take the voice ports for a specific VIC offline, use the **shutdown** voice-port configuration command. 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.

Default

Enabled.

Command Mode

Voice-port configuration.

Usage Guidelines

When you enter the **shutdown** command, all ports on the VIC are disabled, and there is dead silence on the telephone connected to the interface. When you enter the **no shutdown** command, all ports on the VIC are enabled.

Example

The following example takes voice port 1/0 offline:

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

Note The preceding configuration example first shuts down voice port 1/0 and then voice port 1/1.

signal

To specify the type of signaling for a voice port, use the **signal** voice-port configuration command. Use the **no** form of this command to restore the default value for this command.

signal {**loop-start** | **ground-start** | **wink-start** | **immediate** | **delay-dial**}
no signal

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Syntax Description

loop-start	Loop Start signaling. Used for FXO and FXS interfaces. With Loop Start signaling, only one side of a connection can hang up. This is the default setting for FXO and FXS voice ports.
ground-start	Ground Start signaling. Used for FXO and FXS interfaces. Ground Start allows both sides of a connection to place a call and to hang up.
wink-start	Calling side seizes the line by going off-hook on its E lead and 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	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	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.

Default

loop-start for FXO and FXS interfaces.

wink-start for E&M interfaces.

Command Mode

Voice-port configuration.

Usage Guidelines

Configuring the **signal** command for an FXS or FXO voice port changes the signal value for both voice ports on a VIC.

Note If you change the signal type for an FXO voice port, you need to move the appropriate jumper in the VIC.

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 take effect.

Some PBXs miss initial digits if the E&M voice port is configured for **immediate** signaling. If this occurs, use **delay-dial** signaling instead. Some devices (not Cisco devices) have a limited number of DTMF receivers. This type of equipment must delay the calling side until a DTMF receiver is available.

Example

The following example configures **ground-start** signaling, which means that both sides of a connection can place a call and hang up, as the signaling type for a voice port:

```
configure terminal
 voice port 1/1
 signal ground-start
```

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** dial-peer configuration command. Use the **no** form of this command to disable this feature.

snmp enable peer-trap poor-qov
no snmp enable peer-trap poor-qov

Syntax Description

This command has no arguments or keywords.

Default

Disabled.

Command Mode

Dial-peer configuration.

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 uses SNMP messages when voice quality drops, you might want to enable this command. Otherwise, you should disable this command to reduce unnecessary network traffic.

This command only applies to VoIP peers.

Example

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

snmp-server enable traps voice poor-qov
snmp trap link-status

snmp-server enable traps

To enable the router to send SNMP traps, use the **snmp-server enable traps** global configuration command. Use the **no** form of this command to disable SNMP traps.

snmp-server enable traps [*trap-type*] [*trap-option*]
no snmp-server enable traps [*trap-type*] [*trap-option*]

Syntax Description

trap-type	(Optional) Type of trap to enable. If no type is specified, all traps are sent (including the **envmon** and **repeater** traps). The trap type can be one of the following keywords: • **bgp**—Sends Border Gateway Protocol (BGP) state change traps. • **config**—Sends configuration traps. • **entity**—Sends Entity Management Information Base (MIB) modification traps. • **envmon**—Sends Cisco enterprise-specific environmental monitor traps when an environmental threshold is exceeded. When the **envmon** keyword is used, you can specify a *trap-option* value. • **frame-relay**—Sends Frame Relay traps. • **isdn**—Sends Integrated Services Digital Network (ISDN) traps. When the **isdn** keyword is used, you can specify a *trap-option* value. • **repeater**—Sends Ethernet hub repeater traps. When the **repeater** keyword is selected, you can specify a *trap-option* value. • **rtr**—Sends response time reporter (RTR) traps. • **snmp**—Sends SNMP traps. When the **snmp** keyword is used, you can specify a *trap-option* value. • **syslog**—Sends error message traps (Cisco Syslog MIB). Specify the level of messages to be sent with the **logging history level** command. • **voice**—Sends SNMP poor-quality-of-voice traps when used with the **qov** *trap-option*.
trap-option	(Optional) When the **envmon** keyword is used, you can enable a specific environmental trap type or accept all trap types from the environmental monitor system. If no option is specified, all environmental types are enabled. The option can be one or more of the following keywords: **voltage**, **shutdown**, **supply**, **fan**, and **temperature**. When the **isdn** keyword is used, you can specify the **call-information** keyword to enable an SNMP ISDN call information trap for the ISDN MIB subsystem, or you can specify the **isdnu-interface** keyword to enable an SNMP ISDN U interface trap for the ISDN U interface MIB subsystem. When the **repeater** keyword is used, you can specify the repeater option. If no option is specified, all repeater types are enabled. The option can be one or both of the following keywords: • **health**—Enables Internet Engineering Task Force (IETF) Repeater Hub MIB (RFC 1516) health trap. • **reset**—Enables IETF Repeater Hub MIB (RFC 1516) reset trap. When the **snmp** keyword is used, you can specify the **authentication** option to enable SNMP Authentication Failure traps. (The **snmp-server enable traps snmp authentication** command replaces the **snmp-server trap-authentication** command.) If no option is specified, all SNMP traps are enabled. When the **voice** keyword is used, you can enable SNMP poor-quality-of-voice traps by using the **qov** option.

Defaults

No traps are enabled.

Some trap types cannot be controlled with this command. These traps are either always enabled or enabled by some other means. For example, the linkUpDown messages are disabled by the **no snmp trap link-status** command.

If you enter this command with no keywords, the default is to enable all trap types.

Command Mode

Global configuration.

Usage Guidelines

This command is useful for disabling traps that are generating a large amount of uninteresting or useless noise.

If you do not enter an **snmp-server enable traps** command, no traps controlled by this command are sent. To configure the router to send these SNMP traps, you must enter at least one **snmp-server enable traps** command. If you enter the command with no keywords, all trap types are enabled. If you enter the command with a keyword, only the trap type related to that keyword is enabled. To enable multiple types of traps, you must issue a separate **snmp-server enable traps** command for each trap type and option.

The **snmp-server enable traps** command is used in conjunction with the **snmp-server host** command. Use the **snmp-server host** command to specify which host or hosts receive SNMP traps. In order to send traps, you must configure at least one **snmp-server host** command.

For a host to receive a trap controlled by this command, both the **snmp-server enable traps** command and the **snmp-server host** command for that host must be enabled. If the trap type is not controlled by this command, just the appropriate **snmp-server host** command must be enabled.

The trap types used in this command all have an associated MIB object that allows them to be globally enabled or disabled. Not all of the trap types available in the **snmp-server host** command have notificationEnable MIB objects, so some of these cannot be controlled using the **snmp-server enable traps** command.

Examples

The following example enables the router to send SNMP poor-quality-of-voice traps:

```
configure terminal
 snmp-server enable trap voice poor-qov
```

The following example enables the router to send all traps to the host myhost.cisco.com using the community string *public*:

```
snmp-server enable traps
snmp-server host myhost.cisco.com public
```

The following example enables the router to send Frame Relay and environmental monitor traps to the host myhost.cisco.com using the community string *public*:

```
snmp-server enable traps frame-relay
snmp-server enable traps envmon temperature
snmp-server host myhost.cisco.com public
```

The following example does not send traps to any host. The BGP traps are enabled for all hosts, but the only traps enabled to be sent to a host are ISDN traps.

```
snmp-server enable traps bgp
snmp-server host bob public isdn
```

Related Commands

snmp enable peer-trap peer-qov
snmp-server host
snmp-server trap-source
snmp trap illegal-address
snmp trap link-status

snmp trap link-status

To enable SNMP trap messages to be generated when this voice port is brought up or down, use the **snmp trap link-status** voice-port configuration command. Use the **no** form of this command to disable this feature.

snmp trap link-status
no snmp trap link-status

Syntax Description

This command contains no arguments or keywords.

Default

Enabled.

Command Mode

Voice-port configuration.

Usage Guidelines

Use the **snmp trap link-status** command to enable SNMP trap messages (linkup and linkdown) to be generated whenever this voice port is brought online or offline.

If you are managing the equipment with an SNMP manager (such as Maestro), enable this command. Enabling **link-status** messages allows the SNMP manager to learn of a status change without polling the equipment. If you are not using an SNMP manager, disable this command to avoid unnecessary network traffic.

Example

The following example enables SNMP trap messages for voice port 1/0:

```
voice port 1/0
 snmp trap link-status
```

Related Commands

snmp enable peer-trap poor-qov
snmp-server enable traps poor-qov

timeouts initial

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

timeouts initial *seconds*
no timeouts initial *seconds*

Syntax Description

seconds	Initial timeout duration in seconds. Valid entries are any integer from 0 to 120.

Default

10 seconds.

Command Mode

Voice-port configuration.

Usage Guidelines

Use the **timeouts initial** command to specify the number of seconds the system waits for the caller to enter the first digit of the dialed digits. The timeouts initial timer is activated when the call is accepted and is deactivated when the caller enters 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.

Example

The following example sets the initial digit timeout value to 15 seconds:

```
voice port 0/0
 timeouts initial 15
```

Related Commands

timeouts interdigit
timing

timeouts interdigit

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

timeouts interdigit *seconds*
no timeouts interdigit *seconds*

Syntax Description

seconds	Interdigit timeout duration in seconds. Valid entries are any integer from 0 to 120.

Default

10 seconds.

Command Mode

Voice-port configuration.

Usage Guidelines

Use the **timeouts interdigit** command to specify the number of seconds the system waits (after the caller has entered the initial digit) for the caller to enter a subsequent digit of the dialed digits. The timeouts interdigit timer is activated when the caller enters a digit and is restarted each time the caller enters 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.

Example

The following example sets the interdigit timeout value to 15 seconds:

```
voice port 0/0
 timeouts interdigit 15
```

Related Commands

timeouts initial
timing

timing

To specify timing parameters (other than those defined by the **timeouts** commands) for a specified voice port, use the **timing** voice-port configuration command. Use the **no** form of this command to reset the default value for this command.

timing *timing-value*
no timing *timing-value*

Syntax Description

timing-value	One of the keyword/argument pairs listed in Table 4-10.

Table 4-10 Timing Keywords/Arguments, Descriptions, and Valid Entries

Keyword/Argument	Argument Description	Valid Entries
clear-wait *milliseconds*	The minimum amount of time, in milliseconds, between the inactive seizure signal and the call being cleared	Numbers from 200 to 2000
delay-duration *milliseconds*	The delay signal duration for delay dial signaling, in milliseconds	Numbers from 100 to 5000
delay-start *milliseconds*	The minimum delay time, in milliseconds, from outgoing seizure to outdial address	Numbers from 20 to 2000
dial-pulse min-delay *milliseconds*	The time, in milliseconds, between the generation of wink-like pulses	Numbers from 0 to 5000
digit *milliseconds*	The DTMF digit signal duration, in milliseconds	Numbers from 50 to 100
inter-digit *milliseconds*	The DTMF inter-digit duration, in milliseconds	Numbers from 50 to 500
pulse *pulses per second*	The pulse dialing rate, in pulses per second	Numbers from 10 to 20
pulse-inter-digit *milliseconds*	The pulse dialing inter-digit timing, in milliseconds	Numbers from 100 to 1000
wink-duration *milliseconds*	The maximum wink signal duration, in milliseconds, for a wink start signal	Numbers from 100 to 400
wink-wait *milliseconds*	The maximum wink-wait duration, in milliseconds, for a wink start signal	Numbers from 100 to 5000

Default

The default values for the **timing** keywords/arguments are listed in Table 4-11.

Table 4-11 Timing Keywords/Arguments Default Values

Keyword/Argument	Default Value
clear-wait *milliseconds*	400 ms

Table 4-11 Timing Keywords/Arguments Default Values

delay-duration *milliseconds*	2000 ms
delay-start *milliseconds*	300 ms
dial-pulse min-delay *milliseconds*	140 ms
digit *milliseconds*	100 ms
inter-digit *milliseconds*	100 ms
pulse *pulses per second*	20 pps
pulse-inter-digit *milliseconds*	500 ms
wink-duration *milliseconds*	200 ms
wink-wait *milliseconds*	200 ms

Command Mode

Voice-port configuration.

Usage Guidelines

Use the **timing** command to specify timing parameters other than those defined by the **timeouts** commands.

Use the **timing** command with the **dial-pulse min-delay** keyword with PBXs requiring a wink-like pulse, even though they have been configured for delay-dial signaling. If the value for this keyword is set to 0, the router does not generate this wink-like pulse.

Table 4-12 lists the call signal directions for the **timing** keyword/argument pairs.

Table 4-12 Timing Keywords/Arguments Call Signal Directions

Timing Keyword/Argument	Call Signal Direction
clear-wait *milliseconds*	Not applicable
delay-duration *milliseconds*	Out
delay-start *milliseconds*	Out
dial-pulse min-delay *milliseconds*	In
digit *milliseconds*	Out
inter-digit *milliseconds*	Out
pulse *pulses per second*	Out
pulse-inter-digit *milliseconds*	Out
wink-duration *milliseconds*	Out
wink-wait *milliseconds*	Out

Example

The following example configures the clear-wait duration to 300 milliseconds:

```
voice port 0/0
 timing clear-wait 300
```

Related Commands

timeouts initial
timeouts interdigit

type

To specify the E&M interface type, use the **type** voice-port configuration command. Use the **no** form of this command to reset the default value for this command.

type {**1** | **2** | **3** | **5**}
no type

Syntax Description

1	For the following lead configuration: E—Output, relay to ground. M—Input, referenced to ground.
2	For 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	For the following lead configuration: E—Output, relay to ground. M—Input, referenced to ground. SB—Connected to –48V. SG—Connected to ground.
5	For the following lead configuration: E—Output, relay to ground. M—Input, referenced to –48V.

Default

1.

Command Mode

Voice-port configuration.

Usage Guidelines

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 by 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 tie-line equipment generates the E-signal to the PBX by connecting it to SG. The M-signal is detected by the PBX connecting it to SB. Although 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.

With **3**, the interface operates the same as type 1 interfaces with respect to the E-signal. However, the M-signal is detected 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 generates the E-signal to the PBX by grounding the E-lead. The PBX detects 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.

Example

The following example selects type 3 as the interface type for your voice port:

```
voice port 0/0
 type 3
```

vad

To enable voice activity detection (VAD) for the calls using this dial peer, use the **vad** dial-peer configuration command. Use the **no** form of this command to disable this feature.

vad
no vad

Syntax Description

This command has no arguments or keywords.

Default

Enabled.

Command Mode

Dial-peer configuration.

Usage Guidelines

Use the **vad** command to enable VAD. With VAD, silence is not transmitted 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 transmitted to the IP backbone.

This command only applies to VoIP peers.

Example

The following example enables VAD:

```
dial-peer voice 10 voip
 vad
```

Related Commands

comfort-noise

voice-port

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

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

Syntax Description

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	Voice port. Valid entries are 0 or 1.

Default

No voice-port mode is configured.

Command Mode

Global configuration.

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.

Example

The following example accesses the voice port configuration mode for a VIC installed in port 0, slot 0:

```
configure terminal
 voice port 0/0
```

Related Commands

dial-peer