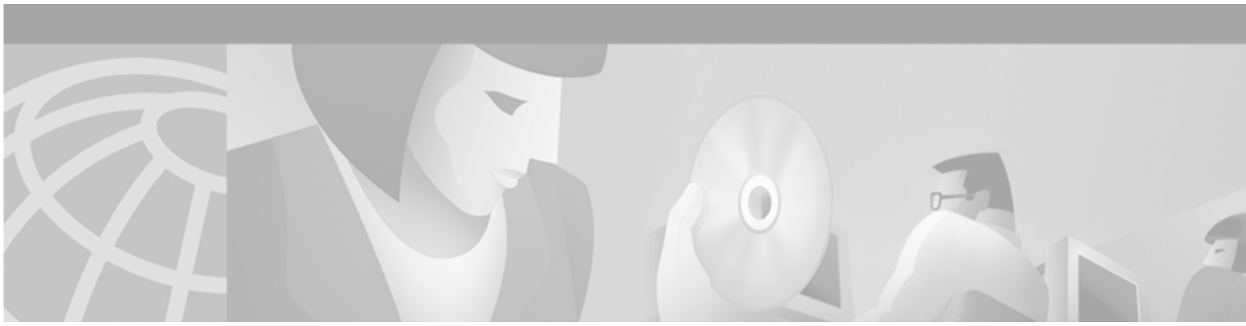

Cisco IOS Voice, Video, and Fax Commands: Si Through Z

This chapter presents the commands to configure and maintain Cisco IOS voice, video, and fax applications. The commands are presented in alphabetical order beginning with Si. Some commands required for configuring voice, video, and fax may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

For detailed information on how to configure these applications and features, refer to the *Cisco IOS Voice, Video, and Fax Configuration Guide*.

signal

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

FXO and FXS Voice Ports

signal {**loop-start** | **ground-start**}

no signal {**loop-start** | **ground-start**}

E&M Voice Ports

signal {**wink-start** | **immediate** | **delay-dial**}

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

Syntax Description

loop-start	Specifies loop start signaling. Used for Foreign Exchange Office (FXO) and Foreign Exchange Station (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	Specifies ground start signaling. Used for FXO and FXS interfaces. Ground start signalling allows both sides of a connection to place a call and to hang up.
wink-start	Indicates that the calling side seizes the line by going off-hook on its E-lead then waits for a short off-hook "wink" indication on its M-lead from the called side before sending address information as dual tone multifrequency (DTMF) digits. Used for E&M tie trunk interfaces. This is the default setting for E&M voice ports.
immediate	Indicates that the calling side seizes the line by going off-hook on its E-lead and sends address information as DTMF digits. Used for E&M tie trunk interfaces.
delay-dial	Indicates that the calling side seizes the line by going off-hook on its E-lead. After a timing interval, the calling side looks at the supervision from the called side. If the supervision is on-hook, the calling side starts sending information as DTMF digits; otherwise, the calling side waits until the called side goes on-hook and then starts sending address information. Used for E&M tie trunk interfaces.

Defaults

Loop-start for FXO and FXS interfaces; wink-start for E&M interfaces.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series routers.

Usage Guidelines

This command applies to analog voice ports only.

Configuring the **signal** command for an FXO or FXS voice port changes the signal value for both voice ports on a voice port module (VPM) card.

Note If you change the signal type for an FXO voice port on Cisco 3600 series routers, you need to move the appropriate jumper in the voice interface card of the voice network module. For more information about the physical characteristics of the voice network module, refer to the installation documentation, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your voice network module.

Configuring this command for an E&M voice port changes only the signal value for the selected voice port. In either case, the voice port must be shut down and then activated before the configured values will take effect.

Some PBXs will miss initial digits if the E&M voice port is configured for Immediate signaling. If this occurs, use Delay-Dial signaling instead. Some non-Cisco devices have a limited number of DTMF receivers. This type of equipment must delay the calling side until a DTMF receiver is available.

Examples

The following example configures ground start signaling on the Cisco 3600 series as the signaling type for a voice port, which means that both sides of a connection can place a call and hang up:

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

signal keepalive

To configure the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks, use the **signal keepalive** command in voice-class configuration mode. To restore the default value, use the **no** form of this command.

signal keepalive *number*

no signal keepalive *number*

Syntax Description

number	Specifies the keepalive signaling packet interval, in seconds. The valid range is from 1 to 65,535.

Defaults

A keepalive packet is sent every 5 seconds.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Before configuring the keepalive signaling interval, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. The voice class must then be assigned to a dial peer.

Examples

The following example, beginning in global configuration mode, sets the keepalive signaling interval to 3 seconds for voice class 10.

```
voice class permanent 10
 signal keepalive 3
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies a dial-peer type.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal timing oos	Configures the signal timing parameter for the OOS state of a call.
voice-class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice class permanent	Assigns a previously-configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal pattern

To define the ABCD bit patterns that identify the idle and out-of-service (OOS) states for Cisco trunks and FRF.11 trunks, use the **signal pattern** command in voice-class configuration mode. To remove the signal pattern setting from the voice class, use the **no** form of this command.

signal pattern {**idle receive** | **idle transmit** | **oos receive** | **oos transmit**} *bit-pattern*

no signal pattern {**idle receive** | **idle transmit** | **oos receive** | **oos transmit**} *bit-pattern*

Syntax Description

idle receive	Defines the signaling pattern for identifying an idle message from the network. Also defines the idle signaling pattern to be sent to the PBX if the network trunk is out of service and the **signal sequence oos idle-only** or **signal sequence oos both** command is configured.
idle transmit	Defines the signaling pattern for identifying an idle message from the PBX.
oos receive	Defines the OOS signaling pattern to be sent to the PBX if the network trunk is out of service and the **signal sequence oos oos-only** or **signal sequence oos both** command is configured.
oos transmit	Defines the signaling pattern for identifying an OOS message from the PBX.
bit-pattern	Defines the ABCD bit pattern. Valid values are from 0000 to 1111.

Defaults

idle receive	For near-end E&M—0000 (for T1) or 0001 (for E1) For near-end FXO loop start—0101 For near-end FXO ground start—1111 For near-end FXS—0101 For near-end MELCAS—1101
idle transmit	For near-end E&M—0000 For near-end FXO—0101 For near-end FXS loop start—0101 For near-end FXS ground start—1111 For near-end MELCAS—1101
oos receive	For near-end E&M—1111 For near-end FXO loop start—1111 For near-end FXO ground start—0000 For near-end FXS loop start—1111 For near-end FXS ground start—0101 For near-end MELCAS—1111
oos transmit	No default signaling pattern is defined.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.0(7)XK	Default signaling patterns were defined.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Before configuring the signaling pattern, you must use the **voice-class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. After you define the voice class, you assign it to a dial peer.

Idle Patterns

An idle state is generated if the router detects an idle signaling pattern coming from either direction. If an idle pattern is configured for only one direction (transmit or receive), an idle state can be detected only in the configured direction. Therefore, you should normally enter both the **idle receive** and the **idle transmit** keywords.

To suppress voice packets whenever the transmit or receive trunk is in the idle state, use the **idle receive** and **idle transmit** keywords in conjunction with the **signal timing idle suppress-voice** command.

OOS Patterns

An OOS state is generated differently in each direction under the following conditions:

- If the router detects an **oos transmit** signaling pattern sent from the PBX, the router transmits the **oos transmit** signaling pattern to the network.
- If the **signal timing oos timeout** timer expires and the router receives no signaling packets from the network (network is OOS), the router sends an **oos receive** signaling pattern to the PBX. (The **oos receive** pattern is not matched against the signaling packets received from the network; the receive packets indicate an OOS condition directly by setting the AIS alarm indication bit in the packet.)

To suppress voice packets whenever the transmit or receive trunk is in the OOS state, use the **oos receive** and **oos transmit** keywords in conjunction with the **signal timing oos suppress-voice** command.

To suppress voice and signaling packets whenever the transmit or receive trunk is in the OOS state, use the **oos receive** and **oos transmit** keywords in conjunction with the **signal timing oos suppress-all** command.

PBX Busyout

To “busy out” a PBX if the network connection fails, set the **oos receive** pattern to match the seized state (busy), and set the **signal timing oos** timeout value. When the timeout value expires and no signaling packets have been received, the router will send the **oos receive** pattern to the PBX.

Use the busy seized pattern only if the PBX does not have a specified pattern for indicating an OOS state. If the PBX has a specific OOS pattern, use that pattern instead.

Examples

The following example, beginning in global configuration mode, configures the signaling bit pattern for the idle receive and transmit states:

```
voice class permanent 10
 signal keepalive 3
 signal pattern idle receive 0101
 signal pattern idle transmit 0101
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

The following example, beginning in global configuration mode, configures the signaling bit pattern for the out-of-service receive and transmit states:

```
voice class permanent 10
 signal keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

The following example restores default signaling bit patterns for the receive and transmit idle states:

```
voice class permanent 10
 signal keepalive 3
 signal timing idle suppress-voice
 no signal pattern idle receive
 no signal pattern idle transmit
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

The following example configures non-default signaling bit patterns for the receive and transmit out-of-service states:

```
voice class permanent 10
 signal keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies a dial-peer type.
signal timing idle suppress-voice	Specifies the length of time before voice traffic is stopped after a trunk goes into the idle state.
signal timing oos	Configures the signal timing parameter for the OOS call state.
signal timing oos slave-standby	Specifies that a slave port return to its initial standby state after the trunk has been OOS for a specified time.
signal timing oos suppress-all	Stops sending voice and signaling packets to the network if a transmit OOS signaling pattern id detected from the PBX for a specified time.
signal timing oos suppress-voice	Stops sending voice packets to the network if a transmit OOS signaling pattern is detected from the PBX for a specified time.

Command	Description
signal timing oos timeout	Changes the delay time between the loss of signaling packets from the network and the start time for the OOS state.
voice-class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice class permanent	Assigns a previously-configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal sequence oos

To specify which signaling pattern is sent to the PBX when the far-end keepalive message is lost or an alarm indication signal (AIS) is received from the far end, use the **signal sequence oos** command in the voice-class configuration mode. To restore the default value, use the **no** form of this command.

signal sequence oos {**no-action** | **idle-only** | **oos-only** | **both**}

no signal sequence oos

Syntax Description

no-action	No signaling pattern is sent.
idle-only	Only the idle signaling pattern is sent.
oos-only	Only the out-of-service (OOS) signaling pattern is sent.
both	Both idle and OOS signaling patterns are sent. This is the default value.

Defaults

Both idle and OOS signaling patterns are sent.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco MC3810 multiservice concentrators.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Before configuring the idle or OOS signaling patterns to be sent, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. After you finish defining the voice class, you assign it to a dial peer.

Use the **signal sequence oos** command to specify which signaling pattern) to send. Use the **signal pattern idle receive** or the **signal pattern oos receive** command to define the bit patterns of the signaling patterns if other than the defaults.

Examples

The following example, beginning in global configuration mode, defines voice class 10, sets the **signal sequence oos** command to send only the idle signalin pattern to the PBX, and applies the voice class configuration to VoFR dial peer 100.

```
voice-class permanent 10
 signal-keepalive 3
 signal sequence oos idle-only
 signal timing idle suppress-voice
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
 signal-type transparent
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies a dial-peer type.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Specifies the length of time before the router stops sending voice packets after a trunk goes into the idle state.
signal timing oos	Specifies that a permanent voice connection be torn down and restarted after the trunk has been OOS for a specified time.
signal timing oos slave-standby	Specifies that a slave port return to its initial standby state after the trunk has been OOS for a specified time.
signal timing oos suppress-all	Configures the router or concentrator to stop sending voice and signaling packets to the network if it detects an OOS signaling pattern from the PBX for a specified time.
signal timing oos suppress-voice	Configures the router or concentrator to stop sending voice packets to the network if it detects a transmit OOS signaling pattern from the PBX for a specified time.
signal timing oos timeout	Changes the delay time between the loss of signaling packets from the network and the start time for the OOS state.
voice-class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice class permanent	Assigns a previously-configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing idle suppress-voice

To configure the signal timing parameter for the idle state of the call, use the **signal timing idle suppress-voice** command in voice-class configuration mode. To restore the default value, use the **no** form of this command.

signal timing idle suppress-voice *seconds*

no signal timing idle suppress-voice *seconds*

Syntax Description

seconds	Duration of the idle state, in seconds, before the voice traffic is stopped. The valid range is from 0 to 65,535.

Defaults

No signal timing idle suppress-voice timer is configured.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.0(7)XK	This command was modified to simplify the configuration process.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Before configuring the signal timing idle suppress-voice timer, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. The voice class must then be assigned to a dial peer.

The **signal timing idle suppress-voice** command is used when the **signal-type** command is set to **transparent** in the dial peer for the Cisco trunk or FRF.11 trunk connection. The router stops sending voice packets when the timer expires. Signaling packets are still sent.

To detect an idle trunk state, the router or concentrator monitors both transmit and receive signaling for the **idle transmit** and **idle receive** signaling patterns. These can be configured by the **signal pattern idle transmit** or **signal pattern idle receive** command, or they can be the defaults. The default **idle receive** pattern is the idle pattern of the local voice port. The default **idle transmit** pattern is the idle pattern of the far-end voice port.

Examples

The following example, beginning in global configuration mode, sets the signal timing idle suppress-voice timer to 5 seconds for the idle state on voice class 10.

```
voice class permanent 10
 signal keepalive 3
 signal pattern idle receive 0101
 signal pattern idle transmit 0101
 signal timing idle suppress-voice 5
 exit
 dial-peer voice 100 vofr
 voice-class permanent 10
 signal-type transparent
```

The following example defines voice class 10, sets the idle detection time to 5 seconds, configures the trunk to use the default transmit and receive idle signal patterns, and applies the voice class configuration to VoFR dial peer 100.

```
voice class permanent 10
 signal keepalive 3
 signal timing idle suppress-voice 5
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
 signal-type transparent
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Defines the ABCD bit patterns that identify the idle and oos states for Cisco trunks and FRF.11 trunks.
signal timing oos	Configures the signal timing parameter for the OOS state of a call.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice-class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice class permanent (dial peer)	Assigns a previously-configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing oos

To configure the signal timing parameter for the out-of-service (OOS) state of the call, use the **signal timing oos** command in voice-class configuration mode. To restore the default value, use the **no** form of this command.

signal timing oos {**restart** | **slave-standby** | **suppress-all** | **suppress-voice** | **timeout**} *seconds*

no signal timing oos {**restart** | **slave-standby** | **suppress-all** | **suppress-voice** | **timeout**} *seconds*

Syntax Description

restart	If no signaling packets are received for this period, the permanent voice connection will be torn down and an attempt to achieve reconnection will be made.
slave-standby	If no signaling packets are received for this period, a slave port returns to its initial standby state. This option applies only to slave ports (ports configured using the **connection trunk** *number* **answer-mode** command).
suppress-all	If the transmit OOS pattern (from the PBX to the network) matches for this period of time, the router stops sending all packets to the network.
suppress-voice	If the transmit OOS pattern (from the PBX to the network) matches for this period of time, the router stops sending voice packets to the network. signaling packets continue to be sent with the alarm indication set (AIS).
timeout	If no signaling packets are received for this period of time, the router sends the configured receive OOS pattern to the PBX. Also, the router stops sending voice packets to the network. Use this option to perform busyout to the PBX.
seconds	Duration, in seconds, for the above settings. The valid range is from 0 to 65,535.

Defaults

No signal timing OOS pattern parameters are configured.

Command Modes

Voice-class configuration

Command History

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

Usage Guidelines

Before configuring signal timing OOS parameters, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. The voice class must then be assigned to a dial peer.

You can enter several values for this command. However, the **suppress-all** and **suppress-voice** options are mutually exclusive.

Examples

The following example, beginning in global configuration mode, configures the signal timeout parameter for the OOS state on voice class 10. The **signal timing oos timeout** command is set to 60 seconds.

```
voice-class permanent 10
 signal-keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 signal timing oos timeout 60
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
connection	Specifies a connection mode for a voice port.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Defines the ABCD bit patterns that identify the idle and oos states for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of the call.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice-class permanent (dial-peer)	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing oos restart

To specify that a permanent voice connection be torn down and restarted after the trunk has been out-of-service (OOS) for a specified time, use the **signal timing oos restart** command in voice-class configuration mode. To restore the default value, use the **no** form of this command.

signal timing oos restart *seconds*

no signal timing oos restart

Syntax Description

seconds	Delay duration, in seconds, for the restart attempt. There is no default duration. The range is from 0 to 65,535.

Defaults

No restart attempt is made if the trunk becomes OOS.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 series.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Before configuring signal timing OOS parameters, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. You then assign the voice class to a dial peer.

The **signal timing oos restart** command is valid only if the **signal timing oos timeout** command is enabled, which controls the start time for the OOS state. The timer for the **signal timing oos restart** command does not start until the trunk is OOS.

Examples

The following example, beginning in global configuration mode, creates voice class 10, sets the OOS timeout time to 60 seconds and sets the restart time to 30 seconds:

```
voice-class permanent 10
 signal-keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 signal timing oos timeout 60
 signal timing oos restart 30
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
connection	Specifies a connection mode for a voice port.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Defines the ABCD bit patterns that identify the idle and oos states for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice-class permanent (dial-peer)	Assigns a previously-configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing oos slave-standby

To configure a slave port to return to its initial standby state after the trunk has been out-of-service (OOS) for a specified time, use the **signal timing oos slave-standby** command in voice-class configuration mode. To restore the default value, use the **no** form of this command.

signal timing oos slave-standby *seconds*

no signal timing oos slave-standby

Syntax Description

seconds	Delay duration, in seconds. If no signaling packets are received for this period, the slave port returns to its initial standby state. There is no default duration. The range is from 0 to 65,535.

Defaults

The slave port does not return to its standby state if the trunk becomes OOS.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 series.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Before configuring signal timing OOS parameters, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. After you finish defining the voice class, you assign it to a dial peer.

If no signaling packets are received for the specified delay period, the slave port returns to its initial standby state. The **signal timing oos slave-standby** command is valid only if both of the following conditions are true:

- The **signal timing oos timeout** command is enabled, which controls the start time for the OOS state. The timer for the **signal timing oos slave-standby** command does not start until the trunk is OOS.
- The voice port is configured as a slave port with the **connection trunk** *digits* **answer-mode** command.

Examples

The following example, beginning in global configuration mode, creates a voice port as a slave voice port, creates voice class 10, sets the OOS timeout time to 60 seconds, and sets the return-to-slave-standby time to 120 seconds:

```
voice-port 1/0/0
 connection trunk 5559262 answer-mode
 exit
 voice-class permanent 10
 signal-keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 signal timing oos timeout 60
 signal timing oos slave-standby 120
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Command History

Command	Description
connection	Specifies a connection mode for a voice port.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Defines the ABCD bit patterns that identify the idle and oos states for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice-class permanent (dial-peer)	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing oos suppress-all

To configure the router or concentrator to stop sending voice and signaling packets to the network if it detects a transmit out-of-service (OOS) signaling pattern from the PBX for a specified time, use the **signal timing oos suppress-all** command in voice-class configuration mode. To restore the default value, use the **no** form of this command.

signal timing oos suppress-all *seconds*

no signal timing oos suppress-all

Syntax Description

seconds	Delay duration, in seconds, before packet transmission is stopped. There is no default duration. The range is from 0 to 65,535.

Defaults

The router or concentrator does not stop sending packets to the network if it detects a transmit OOS signaling pattern from the PBX.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 series.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Before configuring signal timing OOS parameters, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. After you finish defining the voice class, you assign it to a dial peer.

The **signal timing oos suppress-all** command is valid only if you configure an OOS transmit signaling pattern with the **signal pattern oos transmit** command. (There is no default **oos transmit** signaling pattern.)

The **signal timing oos suppress-all** command is valid whether or not the **signal timing oos timeout** command is enabled, which controls the start time for the OOS state. The timer for the **signal timing oos suppress-all** command starts immediately when the OOS transmit signaling pattern is matched.

Examples

The following example, beginning in global configuration mode, creates voice class 10, sets the OOS **timeout** time to 60 seconds, and sets the packet suppression time to 60 seconds:

```
voice-class permanent 10
 signal-keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 signal timing oos timeout 60
 signal timing oos suppress-all 60
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
connection	Specifies a connection mode for a voice port.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Defines the ABCD bit patterns that identify the idle and oos states for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice-class permanent (dial-peer)	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing oos suppress-voice

To configure the router or concentrator to stop sending voice packets to the network if it detects a transmit out-of-service (OOS) signaling pattern from the PBX for a specified time, use the **signal timing oos suppress-voice** command in voice-class configuration mode. To restore the default value, use the **no** form of this command.

signal timing oos suppress-voice *seconds*

no signal timing oos suppress-voice

Syntax Description

seconds	Delay duration, in seconds, before voice-packet transmission is stopped. There is no default duration. The range is from 0 to 65,535.

Defaults

The router or concentrator does not stop sending voice packets to the network if it detects a transmit OOS signaling pattern from the PBX.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 series.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Before configuring signal timing OOS parameters, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. After you finish defining the voice class, you assign it to a dial peer.

The **signal timing oos suppress-voice** command is valid only if you configure an OOS transmit signaling pattern with the **signal pattern oos transmit** command. (There is no default **oos transmit** signaling pattern.)

The **signal timing oos suppress-voice** s command is valid whether or not the **signal timing oos timeout** command is enabled, which controls the start time for the OOS state. The timer for the **signal timing oos suppress-voice** command starts immediately when the OOS transmit signaling pattern is matched.

Examples

The following example, beginning in global configuration mode, creates voice class 10, sets the OOS timeout time to 60 seconds, and sets the packet suppression time to 60 seconds:

```
voice-class permanent 10
 signal-keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 signal timing oos timeout 60
 signal timing oos suppress-voice 60
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
connection	Specifies a connection mode for a voice port.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Defines the ABCD bit patterns that identify the idle and oos states for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice-class permanent (dial-peer)	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing oos timeout

To change the delay time between the loss of signaling packets from the network and the start time for the out-of-service (OOS) state, use the **signal timing oos timeout** command in voice-class configuration mode. To restore the default value, use the **no** form of this command.

signal timing oos timeout [*seconds* | **disabled**]

no signal timing oos timeout

Syntax Description

seconds	(Optional) Delay duration, in seconds, between the loss of signaling packets and the beginning of the OOS state. The default is 30 seconds. The range is from 1 to 65,535.
disabled	(Optional) Deactivates the detection of packet loss. If no signaling packets are received from the network, the router does not sent an OOS pattern to the PBX and it continues sending voice packets to the network. Use this option to disable busyout to the PBX.

Defaults

No signal timing OOS pattern parameters are configured.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 series.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Before configuring signal timing OOS parameters, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. After you finish defining the voice class, you assign it to a dial peer.

You can use the **signal timing oos timeout** command to enable busyout to the PBX.

The **signal timing oos timeout** command controls the starting time for the **signal timing oos restart** and **signal timing oos slave-standby** commands. If this command is entered with the **disabled** keyword, the **signal timing oos restart** and **signal timing oos slave-standby** commands are ineffective.

Examples

The following example, beginning in global configuration mode, creates voice class 10 and sets the OOS timeout time to 60 seconds:

```
voice-class permanent 10
 signal-keepalive 3
 signal pattern oos receive 0001
 signal pattern oos transmit 0001
 signal timing oos timeout 60
 exit
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
connection	Specifies a connection mode for a voice port.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Defines the ABCD bit patterns that identify the idle and oos states for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice-class permanent (dial-peer)	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal-type

To set the signaling type to be used when connecting to a dial peer, use the **signal-type** command in dial-peer configuration mode. To return to the default signal type, use the **no** form of this command.

signal-type {**cas** | **cept** | **ext-signal** | **transparent**}

no signal-type

Syntax Description

cas	North American EIA-464 channel-associated signaling (robbed bit signaling). If the Digital T1 Packet Voice Trunk Network Module is installed, this option might not be available.
cept	Provides a basic E1 ABCD signaling protocol. Used primarily for E&M interfaces. When used with FXS/FXO interfaces, this protocol is equivalent to MELCAS.
ext-signal	External signaling. The digital signal processor (DSP) does not generate any signaling frames. Use this option when there is an external signaling channel, for example, CCS, or when you need to have a permanent "dumb" voice pipe.
transparent	On the Cisco MC3810 multiservice concentrator, selecting this option produces different results depending on whether you are using a digital voice module (DVM) or an analog voice module (AVM). For a DVM: The ABCD signaling bits are copied from or transported through the T1/E1 interface "transparently" without modification or interpretation. This enables the Cisco MC3810 multiservice concentrator to handle arbitrary or unknown signaling protocols. For an AVM: It is not possible to provide "transparent" behavior because the Cisco MC3810 must interpret the signaling information in order to read and write the correct state to the analog hardware. This option is mapped to be equal to **cas**.

Defaults

cas

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrator.
12.0(4)T	Support was added for the Cisco 7200 series router.
12.0(7)XK	In previous releases, the **cept** and **transparent** keywords were supported only on the Cisco MC3810 multiservice concentrator. Beginning in this release, these keywords are supported on the Cisco 2600, 3600 and 7200 series routers.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

This command applies to Voice over Frame Relay (VoFR) and Voice over ATM (VoATM) dial peers. It is used with permanent connections only (Cisco trunks and FRF.11 trunks), not with switched calls.

This command is used to inform the local telephony interface of the type of signaling it should expect to receive from the far-end dial peer. To turn signaling off at this dial peer, select the **ext-signal** option. If signaling is turned off and there are no external signaling channels, a "hot" line exists, enabling this dial peer to connect to anything at the far end.

When you connect an FXS to another FXS, or if you have anything other than an FXS/FXO or E&M/E&M pair, the appropriate signaling type on Cisco 2600 series and 3600 series routers is **ext-signal** (disabled).

If you have a digital E1 connection at the remote end that is running cept/MELCAS signaling and you then trunk that across to an analog port, you should make sure that you configure both ends for the **cept** signal type.

If you have a T1 or E1 connection at both ends and the T1/E1 is running a signaling protocol that is neither EIA-464, or cept/MELCAS, you might want to configure the signal type for the transparent option in order to pass through the signaling.

Examples

The following example shows how to disable signaling on a Cisco 2600 or 3600 series router or on a Cisco MC3810 multiservice concentrator for VoFR dial peer 200, starting from global configuration mode:

```
dial-peer voice 200 vofr
 signal-type ext-signal
 exit
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a dial peer.
connection	Specifies the connection mode for a voice port.
destination-pattern	Specifies the telephone number associated with a dial peer.
dtmf-relay	Enables the DSP to generate FRF.11 Annex A frames for a dial peer.
preference	Enables the preferred dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.
sequence-numbers	Enables the generation of sequence numbers in each frame generated by the DSP.
session protocol	Establishes the VoFR protocol for calls between local and remote routers.
session target	Specifies a network-specific address for a dial peer.

sip-server

To configure a network address for the Session Initiation Protocol (SIP) server interface, use the **sip-server** command in SIP user-agent configuration mode.

sip-server {**dns:**[*host-name*] | **ipv4:***ipaddr*[**:***port-num*]}

Syntax Description

dns:	Sets the global SIP server interface to a Domain Name System (DNS) host name. If you do not specify a host name, the default DNS defined by the **ip name-server** command is used.
host-name	(Optional) A valid DNS host name takes the following format: name.gateway.xyz.
ipv4:*ip_addr*	Sets the global SIP server interface to an IP address. A valid IP address takes the following format: xxx.xxx.xxx.xxx.
:*port-num*	(Optional) Specifies the port number for the SIP server.

Defaults

The default for this command is a null value.

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on Cisco 2600 and 3600 series routers and Cisco AS5300 universal access servers.

Usage Guidelines

If you use this command, you can then specify **session target sip-server** for each dial peer instead of repeatedly entering the SIP server interface address for each dial peer. To reset this command to a null value, use the **default** command.

Examples

The following example, beginning in global configuration mode, sets the global SIP server interface to the DNS host name of UA-1-f0.sip.com:

```
sip-ua
 sip-server dns:UA-1-f0.sip.com
```

Related Commands

Command	Description
sip-ua	Enters SIP user-agent configuration mode, in which you configure the SIP user agent.

sip-ua

To enable the Session Initiation Protocol (SIP) user-agent configuration commands, with which you configure the user agent, use the **sip-ua** command in global configuration mode.

sip-ua

Syntax Description

This command has no arguments or keywords.

Defaults

No default behaviors or values.

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on Cisco 2600 and 3600 series routers and Cisco AS5300 universal access servers.

Usage Guidelines

Use the **sip-ua** command to enter the SIP user-agent configuration mode. Table 71 lists the SIP user-agent configuration mode commands:

Table 71 SIP User-Agent Configuration Mode Commands

Command	Description
exit	Exits SIP user-agent configuration mode.
inband-alerting	Specifies an inband-alerting SIP header.
retry	Configures the SIP signaling timers for retry attempts.
sip-server	Configures a SIP server interface.
timers	Configures the SIP signaling timers configuration.
transport	Enables or disables a SIP user agent transport for TCP or UDP, the protocol SIP user agents will be listening for on port 5060 (default).

Examples

The following example, beginning in global configuration mode, enters SIP user-agent configuration mode, configures the SIP user agent, then returns to global configuration mode:

```
sip-ua
 retry invite 2
 retry response 2
 retry bye 2
 retry cancel 2
 sip-server ipv4:10.0.2.254
 timers invite-wait-100 500
 exit
```

Related Commands

Command	Description
exit	Exits SIP user-agent configuration mode.
inband-alerting	Specifies an inband-alerting SIP header.
retry	Configures the retry attempts for SIP messages.
show sip-ua	Displays statistics for SIP retries, timers, and current listener status.
sip-server	Configures the SIP server interface.
timers	Configures the SIP signaling timers.
transport	Configures the SIP user agent (gateway) for SIP signaling messages on inbound calls through the SIP TCP or UDP socket.

snmp enable peer-trap poor-qov

To generate poor quality of voice notification for applicable calls associated with Voice over IP (VoIP) dial peers, use the **snmp enable peer-trap poor-qov** command in dial-peer configuration mode. To disable this notification, use the **no** form of this command.

snmp enable peer-trap poor-qov

no snmp enable peer-trap poor-qov

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.

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 a Simple Network Management Protocol (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.

Examples

The following example enables poor quality of voice notifications for calls associated with VoIP dial peer 10:

```
dial-peer voice 10 voip
 snmp enable peer-trap poor-qov
```

Related Commands

Command	Description
snmp-server enable traps	Enables a router to send SNMP traps and information.
snmp trap link-status	Enables SNMP trap messages to be generated when a specific port is brought up or down.

ss7 mtp2-variant bellcore

To configure the router for Telcordia Technologies (formerly Bellcore) standards, use the **ss7 mtp2-variant bellcore** command in global configuration mode.

ss7 mtp-variant bellcore [*channel*] [*parameters*]

Syntax Description

channel	Specifies the channel, 0 through 3.
parameters	See table below for timer descriptions, defaults, and ranges.

Defaults

Bellcore is the default variant if no other is configured.

See Table 72 for default parameters.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This MTP2 variant has timers and parameters that can be configured using the values listed in Table 72. To restore the designated default, use the **no** or the **default** form of the command (see example below).

Note Timer durations are converted to 10 millisecond units. For example, a T1 value of 1005 is converted to 100 which results in a actual timeout duration of 1000 milliseconds.
This is true for all timers and all variants.

Table 72 Bellcore (Telcordia Technologies) Parameters and Values

Parameter	Description	Default	Range
T1	aligned/ready timer duration (milliseconds)	13000	1000 to 65535
T2	not aligned timer (milliseconds)	11500	1000 to 65535
T3	aligned timer (milliseconds)	11500	1000 to 65535
T4-Emergency-Proving	emergency proving timer (milliseconds)	600	1000 to 65535
T4-Normal-Proving	normal proving period (milliseconds)	2300	1000 to 65535
T5	sending SIB timer (milliseconds)	100	80 to 65535
T6	remote congestion timer (milliseconds)	6000	1000 to 65535
T7	excessive delay timer (milliseconds)	1000	500 to 65535

Table 72 Bellcore (Telcordia Technologies) Parameters and Values (continued)

Parameter	Description	Default	Range
lssu-len	1 or 2 byte LSSU format	1	1 to 2
unacked-MSUs	Maximum number of MSUs waiting ACK	127	16 to 127
proving-attempts	Maximum number of attempts to prove alignment	5	3 to 8
SUERM-threshold	SUERM error rate threshold	64	32 to 128
SUERM-number-octets	SUERM octet counting mode	16	8 to 32
SUERM-number-signal-units	signal units (good or bad) needed to dec ERM	256	128 to 512
Tie-AERM-Emergency	AERM emergency error rate threshold	1	1 to 8
Tie-AERM-Normal	AERM normal error rate threshold	4	1 to 8

Examples

The following example sets the aligned/ready timer duration on channel 0 to 30,000 milliseconds:

```
ss7 mtp2-variant Bellcore 0
 T1 30000
```

The following example restores the aligned/ready timer default value of 13,000 milliseconds:

```
ss7 mtp2-variant Bellcore 0
 no T1
```

Related Commands

Command	Description
ss7 mtp2-variant itu	Specifies the mtp2-variant as ITU.
ss7 mtp2-variant ntt	Specifies the mtp2-variant as NTT.
ss7 mtp2-variant ttc	Specifies the mtp2-variant as TTC.

ss7 mtp2-variant itu

To configure the router for ITU (International Telecom United) standards, use the **ss7 mtp2-variant itu** command in global configuration mode.

ss7 mtp-variant itu [*channel*] [*parameters*]

Syntax Description

channel	Specifies the channel, 0 through 3.
parameters	See table below for timer descriptions, defaults, and ranges.

Defaults

Bellcore is the default variant if no other is configured.

See Table 73 for ITU default parameters.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The ITU MTP2 variant has timers and parameters that can be configured using the values listed in Table 73. To restore the designated default, use the **no** or the **default** form of the command (see the example below).

Table 73 ITU (White) Parameters and Values

Parameter	Description	Default	Range
T1	aligned/ready timer duration (milliseconds)	40000	1000 to 65535
T2	not aligned timer (milliseconds)	5000	1000 to 65535
T3	aligned timer (milliseconds)	1000	1000 to 65535
T4-Emergency-Proving	emergency proving timer (milliseconds)	500	1000 to 65535
T4-Normal-Proving	normal proving timer (milliseconds)	8200	1000 to 65535
T5	sending SIB timer (milliseconds)	100	80 to 65535
T6	remote congestion timer (milliseconds)	6000	1000 to 65535
T7	excessive delay timer (milliseconds)	1000	1000 to 65535
lssu-len	1 or 2 byte LSSU format	1	1 to 2
msu-len			

Table 73 ITU (White) Parameters and Values (continued)

Parameter	Description	Default	Range
unacked-MSUs	Maximum number of MSUs waiting ACK	127	16 to 127
proving-attempts	Maximum number of attempts to prove alignment	5	3 to 8
SUERM-threshold	SUERM error rate threshold	64	32 to 128
SUERM-number-octets	SUERM octet counting mode	16	8 to 32
SUERM-number-signal-units	signal units (good or bad) needed to dec ERM	256	128 to 512
Tie-AERM-Emergency	AERM emergency error rate threshold	1	1 to 8
Tin-AERM-Normal	AERM normal error rate threshold	4	1 to 8

Examples

The following example sets the emergency proving period on channel 1 to 10,000 milliseconds:

```
ss7 mtp2-variant itu 1
 t4-Emergency-Proving 10000
```

The following example restores the emergency proving period default value of 5,000 milliseconds:

```
ss7 mtp2-variant itu 1
 default t4-Emergency-Proving
```

Related Commands

Command	Description
ss7 mtp2-variant bellcore	Specifies the mtp2-variant as Bellcore.
ss7 mtp2-variant ntt	Specifies the mtp2-variant as NTT.
ss7 mtp2-variant ttc	Specifies the mtp2-variant as TTC.

ss7 mtp2-variant ntt

To configure the router for NTT (Japan) standards, use the **ss7 mtp2-variant ntt** command in global configuration mode.

ss7 mtp-variant ntt [*channel*] [*parameters*]

Syntax Description

channel	Specifies the channel, 0 through 3.
parameters	See table below for timer descriptions, defaults, and ranges.

Defaults

Bellcore is the default variant if no other is configured.

See Table 74 for NTT default parameters.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The NTT MTP2 variant has timers and parameters that can be configured using the values listed in Table 74. To restore the designated default, use the **no** or the **default** form of the command (see the example below).

Table 74 NTT Parameters and Values

Parameter	Description	Default	Range
T1	aligned/ready timer duration (milliseconds)	15000	1000 to 65535
T2	not aligned timer (milliseconds)	5000	1000 to 65535
T3	aligned timer (milliseconds)	3000	1000 to 65535
T4-Emergency-Proving	emergency proving timer (milliseconds)	3000	1000 to 65535
T5	sending SIB timer (milliseconds)	200	80 to 65535
T6	remote congestion timer (milliseconds)	2000	1000 to 65535
T7	excessive delay timer (milliseconds)	3000	1000 to 65535
TA	SIE interval timer (milliseconds)	20	10 to 500
TF	FISU interval timer (milliseconds)	20	10 to 500
TO	SIO interval timer (milliseconds)	20	10 to 500
TS	SIOS interval timer (milliseconds)	20	10 to 500

Table 74 NTT Parameters and Values (continued)

Parameter	Description	Default	Range
unacked-MSUs	Maximum number of MSUs waiting ACK	40	16 to 40
proving-attempts	Maximum number of attempts to prove alignment	5	3 to 8
SUERM-threshold	SUERM error rate threshold	64	32 to 128
SUERM-number-octets	SUERM octet counting mode	16	8 to 32
SUERM-number-signal-units	signal units (good or bad) needed to dec ERM	256	128 to 512
Tie-AERM-Emergency	AERM emergency error rate threshold	1	1 to 8

Examples

The following example sets the SUERM error rate threshold on channel 2 to 100:

```
ss7 mtp2-variant ntt 2
 SUERM-threshold 100
```

The following example restores the SUERM error rate threshold default value of 64:

```
ss7 mtp2-variant ntt 2
 no SUERM-threshold
```

Related Commands

Command	Description
ss7 mtp2-variant bellcore	Specifies the mtp2-variant as Bellcore.
ss7 mtp2-variant itu	Specifies the mtp2-variant as ITU.
ss7 mtp2-variant ttc	Specifies the mtp2-variant as TTC.

ss7 mtp2-variant ttc

To configure the router for TTC (Japan Telecom) standards, use the **ss7 mtp2-variant ttc** command in global configuration mode.

ss7 mtp-variant ttc [*channel*] [*parameters*]

Syntax Description

channel	Specifies the channel, 0 through 3.
parameters	See table below for timer descriptions, defaults, and ranges.

Defaults

Bellcore is the default variant if no other is configured.

See Table 75 for TTC default parameters.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The TTC MTP2 variant has timers and parameters that can be configured using the values listed in Table 75. To restore the designated default, use the **no** or the **default** form of the command (see the example below).

Table 75 TTC Parameters and Values

Parameter	Description	Default	Range
T1	aligned/ready timer duration (milliseconds)	15000	1000 to 65535
T2	not aligned timer (milliseconds)	5000	1000 to 65535
T3	aligned timer (milliseconds)	3000	1000 to 65535
T4-Emergency-Proving	emergency proving timer (milliseconds)	3000	1000 to 65535
T5	sending SIB timer (milliseconds)	200	80 to 65535
T6	remote congestion timer (milliseconds)	2000	1000 to 65535
T7	excessive delay timer (milliseconds)	3000	1000 to 65535
TA	SIE interval timer (milliseconds)	20	10 to 500
TF	FISU interval timer (milliseconds)	20	10 to 500
TO	SIO interval timer (milliseconds)	20	10 to 500
TS	SIOS interval timer (milliseconds)	20	10 to 500

Table 75 TTC Parameters and Values (continued)

Parameter	Description	Default	Range
unacked-MSUs	Maximum number of MSUs waiting ACK	40	16 to 40
proving-attempts	Maximum number of attempts to prove alignment	5	3 to 8
SUERM-threshold	SUERM error rate threshold	64	32 to 128
SUERM-number-octets	SUERM octet counting mode	16	8 to 32
SUERM-number-signal-units	signal units (good or bad) needed to dec ERM	256	128 to 512
Tie-AERM-Emergency	AERM emergency error rate threshold	1	1 to 8

Examples

The following example sets the maximum number of proving attempts for channel 3 to 3:

```
ss7 mtp2-variant ttc 3
 proving-attempts 3
```

The following example restores the maximum number of proving attempts to the default value:

```
ss7 mtp2-variant ttc 3
 default proving-attempts
```

Related Commands

Command	Description
ss7 mtp2-variant bellcore	Specifies the mtp2-variant as Bellcore.
ss7 mtp2-variant itu	Specifies the mtp2-variant as ITU.
ss7 mtp2-variant ntt	Specifies the mtp2-variant as NTT.

ss7 session

To create a Reliable User Datagram Protocol (RUDP) session, use the **ss7 session** command in global configuration mode. To delete the session, use the **no** form of this command.

ss7 session-*session number* {**address** *remote-address remote-port local-address local-port*}

no ss7session-*session number* **address**

Syntax Description

session-number	SS7 session number. Valid values are 0 and 1. You must enter the hyphen, with no space following it, after the **session** keyword.
remote-address	The remote IP address of the Media Gateway Controller in four-part dotted-decimal format.
remote-port	The number of the remote UDP port on which the Media Gateway Controller is configured to listen. This UDP port cannot be used by another protocol as defined in RFC 1700 and cannot be otherwise used in the network.
local-address	The local IP address of the router in four-part dotted-decimal format. The local IP address for both sessions, 0 and 1, must be the same.
local-port	The number of the local UDP port on which the router expects to receive messages from the Media Gateway Controller. Specify any UDP port that is not used by another protocol as defined in RFC 1700 and that is not otherwise used in your network. The local UDP port must be different for session-0 and session-1.

Defaults

No session is configured.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

You can configure a maximum of two sessions, one for each signaling link. In a redundant Media Gateway Controller configuration, session-0 is configured to one MGC and session-1 is configured to the other.

The Media Gateway Controller must be configured to send messages to the local port, and it must be configured to listen on the remote port.

You must reload the router whenever you remove a session or change the parameters of a session.

Examples

The following example sets up two sessions on a Cisco 2611:

```
ss7 session-0 address 255.251.255.255 7000 255.255.255.254 7000
ss7 session-1 address 255.255.255.252 7002 255.255.255.254 7001
```

Note The example above shows how the local IP addresses in session-0 and session-1 must be the same.

Related Commands

Command	Description
ss7 session retrans_t	Sets the retransmission timer.
ss7 session m_retrans	Sets the maximum number of times that the RUDP attempts to resend a segment before declaring the connection invalid.
ss7 session m_rcvnum	Sets the maximum number of segments that the remote end can send before receiving an acknowledgment.
ss7 session m_outseq	Sets the maximum number of out-of-sequence segments that can be received before the RUDP sends an extended acknowledgment.
ss7 session m_cumack	Sets the maximum number of segments that can be received before the RUDP sends an acknowledgment.
ss7 session k_pt	Sets the null segment (keepalive) timer.
ss7 session cumack_t	Sets the cumulative acknowledgment timer.

ss7 session cumack_t

To set the Reliable User Datagram Protocol (RUDP) cumulative acknowledgment timer for a specific SS7 signaling link session, use the **ss7 session cumack_t** command in global configuration mode. To restore the default value, use the **no** form of this command.

ss7 session-*session number* **cumack_t** *milliseconds*

no ss7 session-*session number* **cumack_t**

Caution Use the default setting. Do not change session timers unless instructed to do so by Cisco technical support. Changing timers may result in service interruption or outage.

Syntax Description

session-number	SS7 session number. Valid values are 0 and 1. You must enter the hyphen, with no space following it, after the **session** keyword.
milliseconds	Use this parameter to specify the amount of time (in milliseconds) that the RUDP waits before it sends an acknowledgment after receiving a segment. Valid values are from 100 to 65535. This value should be less than the value configured for the retransmission timer by using the **ss7 session-***session number* **retrans_t** command.

Defaults

The default value is 300 milliseconds.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The cumulative acknowledgment timer determines when the receiver sends an acknowledgment. If the timer is not already running, it is initialized when a valid data, null, or reset segment is received. When the cumulative acknowledgment timer expires, the last in-sequence segment is acknowledged. The RUDP typically tries to “piggyback” acknowledgments on data segments being sent. However, if no data segment is sent in this period of time, it sends a standalone acknowledgment.

Examples

The following example sets up two sessions and sets the cumulative acknowledgment timer to 320 milliseconds for each one:

```
ss7 session-0 address 255.255.255.251 7000 255.255.255.254 7000
ss7 session-0 cumack_t 320
ss7 session-1 address 255.255.255.253 7002 255.255.255.254 7001
ss7 session-1 cumack_t 320
```

Related Commands

Command	Description
ss7 session retrans_t	Sets the retransmission timer.
ss7 session m_retrans	Sets the maximum number of times that the RUDP attempts to resend a segment before declaring the connection invalid.
ss7 session m_rcvnum	Sets the maximum number of segments that the remote end can send before receiving an acknowledgment.
ss7 session m_outseq	Sets the maximum number of out-of-sequence segments that can be received before the RUDP sends an extended acknowledgment.
ss7 session m_cumack	Sets the maximum number of segments that can be received before the RUDP sends an acknowledgment.
ss7 session k_pt	Sets the null segment (keepalive) timer.
show ss7	Displays the SS7 configuration.

ss7 session kp_t

To set the null segment (keepalive) timer for a specific SS7 signaling link session, use the **ss7 session kp_t** command in global configuration mode. To restore the default value, use the **no** form of this command.

ss7 session-*session number* **kp_t** *milliseconds*

no ss7 session-*session number* **kp_t**

Caution Use the default setting. Do not change session timers unless instructed to do so by Cisco technical support. Changing timers may result in service interruption or outage.

Syntax Description

session-number	SS7 session number. Valid values are 0 and 1. You must enter the hyphen, with no space following it, after the **session** keyword.
milliseconds	Use this parameter to specify the amount of time (in milliseconds) that the Reliable User Datagram Protocol (RUDP) waits before sending a keepalive to verify that the connection is still active. Valid values are 0 and from100 to 65535.

Defaults

The default value is 2000 milliseconds.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The null segment timer determines when a null segment (keepalive) is sent by the client Cisco 2600 series router. On the client, the timer starts when the connection is established and is reset each time a data segment is sent. If the null segment timer expires, the client sends a keepalive to the server to verify that the connection is still functional. On the server, the timer restarts each time a data or null segment is received from the client.

The value of the server's null segment timer is twice the value configured for the client. If no segments are received by the server in this period of time, the connection is no longer valid.

To disable keepalive, set this parameter to 0.

Examples

The following example sets up two sessions and sets a keepalive of 1,800 milliseconds for each one:

```
ss7 session-0 address 255.255.255.251 7000 255.255.255.254 7000
ss7 session-0 kp_t 1800
ss7 session-1 address 255.255.255.253 7002 255.255.255.254 7001
ss7 session-1 kp_t 1800
```

Related Commands

Command	Description
ss7 session retrans_t	Sets the retransmission timer.
ss7 session m_retrans	Sets the maximum number of times that the RUDP attempts to resend a segment before declaring the connection invalid.
ss7 session m_rcvnum	Sets the maximum number of segments that the remote end can send before receiving an acknowledgment.
ss7 session m_outseq	Sets the maximum number of out-of-sequence segments that can be received before the RUDP sends an extended acknowledgment.
ss7 session m_cumack	Sets the maximum number of segments that can be received before the RUDP sends an acknowledgment.
ss7 session cumack_t	Sets the cumulative acknowledgment timer.
show ss7	Displays the SS7 configuration.

ss7 session m_cumack

To set the maximum number of segments that can be received before the Reliable User Datagram Protocol (RUDP) sends an acknowledgment in a specific SS7 signaling link session, use the **ss7 session m_cumack** command in global configuration mode. To restore the default value, use the **no** form of this command.

ss7 session-*session number* **m_cumack** *segments*

no ss7 session-*session number* **m_cumack**

Caution Use the default setting. Do not change session timers unless instructed to do so by Cisco technical support. Changing timers may result in service interruption or outage.

Syntax Description

session-number	SS7 session number. Valid values are 0 and 1. You must enter the hyphen, with no space following it, after the **session** keyword.
segments	Use this parameter to specify maximum number of segments that can be received before the Reliable User Datagram Protocol (RUDP) sends an acknowledgment. Valid values are from 0 to 255.

Defaults

The default value is 3 segments.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The cumulative acknowledgment counter records the number of unacknowledged, in-sequence data, null, or reset segments received without a data, null, or reset segment being sent to the transmitter. If this counter reaches the configured maximum, the receiver sends a standalone acknowledgment (a standalone acknowledgment is a segment that contains only acknowledgment information). The standalone acknowledgment contains the sequence number of the last data, null, or reset segment received.

If you set this parameter to 0, an acknowledgment is sent immediately after a data, null, or reset segment is received.

Examples

The following example sets up two sessions and in each session sets a maximum of two segments for receipt before acknowledgment:

```
ss7 session-0 address 255.255.255.251 7000 255.255.255.254 7001
ss7 session-0 m_cumack 2
ss7 session-1 address 255.255.255.253 7002 255.255.255.254 7000
ss7 session-1 m_cumack 2
```

Related Commands

Command	Description
ss7 session retrans_t	Sets the retransmission timer.
ss7 session m_retrans	Sets the maximum number of times that the RUDP attempts to resend a segment before declaring the connection invalid.
ss7 session m_rcvnum	Sets the maximum number of segments that the remote end can send before receiving an acknowledgment.
ss7 session m_outseq	Sets the maximum number of out-of-sequence segments that can be received before the RUDP sends an extended acknowledgment.
ss7 session k_pt	Sets the null segment (keepalive) timer.
ss7 session cumack_t	Sets the cumulative acknowledgment timer.
show ss7	Displays the SS7 configuration.

ss7 session m_outseq

To set the maximum number of out-of-sequence segments that can be received before the Reliable User Datagram Protocol (RUDP) sends an extended acknowledgment in a specific SS7 signaling link session, use the **ss7 session m_outseq** command in global configuration mode. To restore the default value, use the **no** form of this command.

ss7 session-*session number* **m_outseq** *segments*

no ss7 session-*session number* **m_outseq**

Caution Use the default setting. Do not change session timers unless instructed to do so by Cisco technical support. Changing timers may result in service interruption or outage.

Syntax Description

session-number	SS7 session number. Valid values are 0 and 1. You must enter the hyphen, with no space following it, after the **session** keyword.
segments	Use this parameter to specify the maximum number of out-of-sequence segments that can be received before the RUDP sends an extended acknowledgment. If the specified number of segments are received out of sequence, an Extended Acknowledgment segment is sent to inform the sender which segments are missing. Valid values are from 0 to 255.

Defaults

The default value is 3 segments.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The out-of-sequence acknowledgment counter records the number of data segments that have arrived out of sequence. If this counter reaches the configured maximum, the receiver sends an extended acknowledgment segment that contains the sequence numbers of the out-of-sequence data, null, and reset segments received. When the transmitter receives the extended acknowledgment segment, it retransmits the missing data segments.

If you set this parameter to 0, an acknowledgment is sent immediately after an out-of-sequence segment is received.

Examples

The following example sets up two sessions and sets a maximum number of four out-of-sequence segments for each session:

```
ss7 session-0 address 255.255.255.251 7000 255.255.255.254 7001
ss7 session-0 m_outseq 4
ss7 session-1 address 255.255.255.253 7002 255.255.255.254 7000
ss7 session-1 m_outseq 4
```

Related Commands

Command	Description
ss7 session retrans_t	Sets the retransmission timer.
ss7 session m_retrans	Sets the maximum number of times that the RUDP attempts to resend a segment before declaring the connection invalid.
ss7 session m_rcvnum	Sets the maximum number of segments that the remote end can send before receiving an acknowledgment.
ss7 session m_cumack	Sets the maximum number of segments that can be received before the RUDP sends an acknowledgment.
ss7 session k_pt	Sets the null segment (keepalive) timer.
ss7 session cumack_t	Sets the cumulative acknowledgment timer.
show ss7	Displays the SS7 configuration.

ss7 session m_rcvnum

To set the maximum number of segments that the remote end can send before receiving an acknowledgment in a specific SS7 signaling link session, use the **ss7 session m_rcvnum** command in global configuration mode. To restore the default value, use the **no** form of this command.

ss7 session-*session number* **m_rcvnum** *segments*

no ss7 session-*session number* **m_rcvnum**

Caution Use the default setting. Do not change session timers unless instructed to do so by Cisco technical support. Changing timers may result in service interruption or outage.

Syntax Description

session-number	SS7 session number. Valid values are 0 and 1. You must enter the hyphen, with no space following it, after the **session** keyword.
segments	Use this parameter to specify the maximum number of segments that the remote (Cisco IOS software) end can send before receiving an acknowledgment. Valid values are from 1 to 64.

Defaults

The default value is 32 segments.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The outstanding segments counter is the maximum number of segments that the Cisco IOS software end of the connection can send without getting an acknowledgment from the receiver. The receiver uses the counter for flow control.

Examples

The following example sets up two sessions and for each session sets a maximum of 36 segments for receipt before an acknowledgment:

```
ss7 session-0 address 255.255.255.251 7000 255.255.255.254 7001
ss7 session-0 m_rcvnum 36
ss7 session-1 address 255.255.255.253 7002 255.255.255.254 7000
ss7 session-1 m_rcvnum 36
```

Related Commands

Command	Description
ss7 session retrans_t	Sets the retransmission timer.
ss7 session m_retrans	Sets the maximum number of times that the Reliable User Datagram Protocol (RUDP) attempts to resend a segment before declaring the connection invalid.
ss7 session m_outseq	Sets the maximum number of out-of-sequence segments that can be received before the RUDP sends an extended acknowledgment.
ss7 session m_cumack	Sets the maximum number of segments that can be received before the RUDP sends an acknowledgment.
ss7 session k_pt	Sets the null segment (keepalive) timer.
ss7 session cumack_t	Sets the cumulative acknowledgment timer.
show ss7	Displays the SS7 configuration.

ss7 session m_retrans

To set the maximum number of times that the Reliable User Datagram Protocol (RUDP) attempts to resend a segment before declaring the connection invalid in a specific SS7 signaling link session, use the **ss7 session m_retrans** command in global configuration mode. To restore the default value, use the **no** form of this command.

ss7 session-*session number* **m_retrans** *number*

no ss7 session-*session number* **m_retrans**

Caution Use the default setting. Do not change session timers unless instructed to do so by Cisco technical support. Changing timers may result in service interruption or outage.

Syntax Description

session-number	SS7 session number. Valid values are 0 and 1. You must enter the hyphen, with no space following it, after the **session** keyword.
number	Use this parameter to specify the maximum number of times that the RRUDP attempts to resend a segment before declaring the connection broken. Valid values are from 0 to 255.

Defaults

The default value is 2 times.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The retransmission counter is the number of times a segment has been retransmitted. If this counter reaches the configured maximum, the transmitter resets the connection and informs the upper-layer protocol.

If you set this parameter to 0, the RUDP attempts to resend the segment continuously.

Examples

The following example sets up two sessions and for each session sets a maximum number of three times to resend before a session becomes invalid:

```
ss7 session-0 address 255.255.255.251 7000 255.255.255.254 7001
ss7 session-0 m_retrans 3
ss7 session-1 address 255.255.255.253 7002 255.255.255.254 7000
ss7 session-1 m_retrans 3
```

Related Commands

Command	Description
ss7 session retrans_t	Sets the retransmission timer.
ss7 session m_rcvnum	Sets the maximum number of segments that the remote end can send before receiving an acknowledgment.
ss7 session m_outseq	Sets the maximum number of out-of-sequence segments that can be received before the RUDP sends an extended acknowledgment.
ss7 session m_cumack	Sets the maximum number of segments that can be received before the RUDP sends an acknowledgment.
ss7 session k_pt	Sets the null segment (keepalive) timer.
ss7 session cumack_t	Sets the cumulative acknowledgment timer.
show ss7	Displays the SS7 configuration.

ss7 session retrans_t

To set the amount of time that the Reliable User Datagram Protocol (RUDP) waits to receive an acknowledgment for a segment in a specific SS7 signaling link session, use the **ss7 session retrans_t** command in global configuration mode. If the RUDP does not receive the acknowledgment in this time period, the RUDP retransmits the segment. To restore the default value, use the **no** form of this command.

ss7 session-*session number* **retrans_t** *milliseconds*

no ss7 session-*session number* **retrans_t**

Caution Use the default setting. Do not change session timers unless instructed to do so by Cisco technical support. Changing timers may result in service interruption or outage.

Syntax Description

session-number	SS7 session number. Valid values are 0 and 1. You must enter the hyphen, with no space following it, after the **session** keyword.
milliseconds	Use this parameter to specify the amount of time that the RUDP waits to receive an acknowledgment for a segment. Valid values are from 100 to 65535.

Defaults

The default value is 600 milliseconds.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

The retransmission timer is used to determine whether a packet must be retransmitted and is initialized each time a data, null, or reset segment is sent. If an acknowledgment for the segment is not received by the time the retransmission timer expires, all segments that have been transmitted—but not acknowledged—are retransmitted.

This value should be greater than the value configured for the cumulative acknowledgment timer by using the **ss7 session cumack_t** command.

Examples

The following example sets up two sessions and specifies 550 milliseconds as the time to wait for an acknowledgment for each session:

```
ss7 session-0 address 255.255.255.251 7000 255.255.255.254 7001
ss7 session-0 retrans_t 550
ss7 session-1 address 255.255.255.253 7002 255.255.255.254 7000
ss7 session-1 retrans_t 550
```

Related Commands

Command	Description
ss7 session m_retrans	Sets the maximum number of times that the RUDP attempts to resend a segment before declaring the connection invalid.
ss7 session m_rcvnum	Sets the maximum number of segments that the remote end can send before receiving an acknowledgment.
ss7 session m_outseq	Sets the maximum number of out-of-sequence segments that can be received before the RUDP sends an extended acknowledgment.
ss7 session m_cumack	Sets the maximum number of segments that can be received before the RUDP sends an acknowledgment.
ss7 session k_pt	Sets the null segment (keepalive) timer.
ss7 session cumack_t	Sets the cumulative acknowledgment timer.
show ss7	Displays the SS7 configuration.

ss7 set failover-timer

To specify the amount of time that the SS7 Session Manager waits for the active session to recover or for the standby Media Gateway Controller to indicate that the SLT should switch traffic to the standby session, use the **ss7 set failover-timer** command in global configuration mode. To restore the default setting, use the **no** form of this command.

ss7 set failover-timer [*seconds*]

no ss7 set failover-timer

Syntax Description

seconds	Time in seconds that the Session Manager waits for a session to recover. Values from 1 through 10 are valid.

Defaults

The default is 3 seconds.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR	This command was introduced.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

This command specifies the number of seconds that the Session Manager waits for the the active session to recover or for the standby Media Gateway Controller to indicate that the SLT should switch traffic to the standby session and to make that session the active session. If the timer expires without a recovery of the original session or an active message from the standby Media Gateway Controller, the signaling links are taken out of service.

Examples

The following example sets the failover timer to four seconds:

```
ss7 set failover-timer 4
```

Related Commands

Command	Description
show ss7 sm set	Displays the current failover timer setting.
ss7 session	Establishes a session.

station-id

To specify the name or number that will be sent as Caller-ID information and enable Caller-ID, use the **station-id** voice-port configuration command at the sending Foreign Exchange Station (FXS) voice port or at a Foreign Exchange Office (FXO) port through which routed Caller-ID calls pass. To remove the name or number, use the **no** form of this command.

station-id [name *name* | **number** *number*]

no station-id [name *name* | **number** *number*]

Syntax Description

name	A string of 1 to 15 characters to represent the station name.
number	A string of from 1 to 15 characters to represent the station number.

Defaults

The default is no station name or number.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was implemented for Cisco MC3810 and for Cisco 2600 and 3600 series routers.
12.1(3)T	This command was first supported on the T Train.

Usage Guidelines

This optional command is configured on FXS voice ports that are used to originate on-net calls. The information entered is displayed by the telephone attached to the FXS port at the far end of the on-net call. It can also be configured on the FXO port of a router on which Caller ID information is expected to be received from the CO, to suit situations where a call is placed from the CO, then goes through the FXO interface, and continues to a far-end FXS port through an on-net call. In this case, if no Caller ID information is received from the CO telephone line, the far-end call recipient receives the information configured on the FXO port.

Note This feature applies only to Caller ID name display provided by an FXS port connection to a telephone device. The station name will not be passed through telephone trunk connections supporting Automatic Number Identification (ANI) calls. ANI supplies calling number identification only and does not support calling number names.

Do not use this command when the Caller-ID standard is dual-tone multifrequency (DTMF). DTMF Caller ID can carry only the calling number.

If the **station-id** or **caller-id alerting** command is configured on the voice port, these automatically enable Caller-ID, and the **caller-id enable** command is not necessary.

This command applies to the Cisco MC3810 and to Cisco 2600 and 3600 series routers.

Examples

The following example configures a Cisco 2600 or 3600 series router voice port from which Caller-ID information is sent:

```
voice-port 1/0/1
   cptone US
   station-id name A. Person
   station-id number 4085551111
```

The following example configures a Cisco MC3810 voice port from which Caller-ID information is sent:

```
voice-port 1/0
   cptone northamerica
   station-id name A. Person
   station-id number 4085551111
   caller-id alerting ring 1
```

Related Commands

Command	Description
caller-id enable	Enables Caller-ID operation.

subcell-mux

To enable subcell multiplexing on a Cisco MC3810 multiservice concentrator, use the **subcell-mux** command in voice-service configuration mode. To restore the default value, use the **no** form of the command.

subcell-mux

no subcell-mux

Syntax Description

This command has no arguments or keywords.

Defaults

Subcell multiplexing is not enabled.

Command Modes

Voice-service configuration

Command History

Release	Modification
12.1(1)XA	The command was introduced for the Cisco MC3810 multiservice concentrator.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Use this command to enable ATM adaptation layer 2 (AAL2) common part sublayer (CPS) subcell multiplexing when the Cisco MC3810 interoperates with other equipment that uses subcell multiplexing.

Examples

The following example enables AAL2 CPS subcell multiplexing on a Cisco MC3810 multiservice concentrator:

```
voice service voatm
 session protocol aal2
 subcell-mux
```

supervisory disconnect

To enable a supervisory disconnect signal on Foreign Exchange Office (FXO) ports, use the **supervisory disconnect** command in voice-port configuration mode. To disable the supervisory disconnect signal, use the **no** form of this command.

supervisory disconnect

no supervisory disconnect

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on Cisco MC3810 multiservice concentrators.

Usage Guidelines

This command indicates whether supervisory disconnect signaling is available on the FXO port. Supervisory disconnect signaling is a power denial from the switch lasting at least 350 milliseconds. When this condition is detected, the system interprets this as a disconnect indication from the switch and clears the call.

You should configure no supervisory disconnect on the voice port if there is no supervisory disconnect available from the switch.

Note If there is no disconnect supervision on the voice port, the interface could be left active if the caller abandons the call before the far end answers. After the router collects the dialed digits but before the called party answers, the router starts a tone detector. Within this time window, the tone detector listens for signals (such as a fast busy signal) that occur if the originating caller hangs up. If this occurs, the router will interpret those tones as a disconnect indication and close the window.

Examples

The following example configures supervisory disconnect on a Cisco 3600 series voice port:

```
voice-port 2/1/0
 supervisory disconnect
```

The following example configures supervisory disconnect on a Cisco MC3810 multiservice concentrator voice port:

```
voice-port 1/1
 supervisory disconnect
```

supervisory disconnect anytone

To configure an Foreign Exchange Office (FXO) voice port to go on-hook if the router detects any tone from a PBX or public switched telephone network (PSTN) before the call is answered, use the **supervisory disconnect anytone** command in voice-port configuration mode. To restore the default, use the **no** form of this command.

supervisory disconnect anytone

no supervisory disconnect anytone

Syntax Description

This command has no arguments or keywords.

Defaults

The supervisory disconnect function is not enabled on voice ports.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600, 3600, and MC3810 series.

Usage Guidelines

The **supervisory disconnect anytone** voice-port configuration command can be used to provide the disconnect function if the PBX or PSTN does not provide a supervisory tone. This function is enabled only during call setup (before the call is answered); examples of tones that trigger a disconnect include busy tone, fast busy tone, and dial tone. You must enable echo cancellation; otherwise, the router's own ringback tone can trigger a disconnect.

This command replaces the **no supervisory disconnect signal** command. If you enter the **no supervisory disconnect signal** command, the supervisory disconnect any-tone feature will be enabled, and "supervisory disconnect" anytone will be displayed when **show** commands are entered.

Examples

The following example configures voice ports 1/4 and 1/5 to go on-hook if any tone from the PBX or PSTN is detected before the call is answered:

```
voice-port 1/4
 supervisory disconnect anytone
 exit
voice-port 1/5
 supervisory disconnect anytone
 exit
```

The following example disables the disconnect function on voice port 1/5:

```
voice-port 1/5
 no supervisory disconnect anytone
 exit
```

Related Commands

Command	Description
voice class dualtone	Creates a voice class for FXO tone detection parameters.

supervisory disconnect dualtone voice-class

To assign a previously configured voice class for Foreign Exchange Office (FXO) supervisory disconnect tone to a voice port, use the **supervisory disconnect dualtone voice-class** command in voice port configuration mode. To remove a voice class from a voice-port, use the **no** form of this command.

supervisory disconnect dualtone {**mid-call** | **pre-connect**} **voice-class** *tag*

no supervisory disconnect dualtone voice-class *tag*

Syntax Description

mid-call	Configures tone detection to operate throughout the duration of the call.
pre-connect	Configures tone detection to operate during call setup and to stop when the called telephone goes off-hook.
tag	A unique identification number assigned to one voice class. The tag number maps to the tag number assigned using the **voice class dualtone** global configuration command. The range is from 1 to 10,000.

Defaults

No voice class is assigned to a voice port.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600, 3600, and MC3810 series.

Usage Guidelines

You can apply an FXO supervisory disconnect tone voice class to multiple voice ports. You can assign only one FXO supervisory disconnect tone voice class to a voice port. If a second voice class is assigned to a voice port, the second voice class replaces the one previously assigned. You cannot assign separate FXO supervisory disconnect tone commands directly to the voice port.

This feature is applicable to analog FXO voice ports with loop-start signaling.

Examples

The following example assigns voice class 70 to FXO voice port 1/5 of a Cisco MC3810 series concentrator and specifies tone detection during the entire call duration:

```
voice-port 1/5
 no echo-cancel enable
 supervisory disconnect dualtone mid-call voice-class 70
```

The following example assigns voice class 80 to FXO voice port 0/1/1 of a Cisco 3600 series router and specifies tone detection only during call setup:

```
voice-port 0/1/1
 no echo-cancel enable
 supervisory disconnect dualtone pre-connect voice-class 80
```

Related Commands

Command	Description
channel-group	Defines the time slots of each T1 or E1 circuit.
mode	Sets the mode of the T1/E1 controller and enters specific configuration commands for each mode type in VoATM.
voice class dualtone	Creates a voice class for FXO tone detection parameters.

tdm-group

To configure a list of time slots for creating clear channel groups (pass-through) for time-division multiplexing (TDM) cross-connect, use the **tdm-group** command in controller configuration mode. To delete a clear channel group, use the **no** form of this command.

tdm-group *tdm-group-no* **timeslot** *timeslot-list* [**type** {**e&m** | **fxs** [**loop-start** | **ground-start**] | **fxo** [**loop-start** | **ground-start**] | **fxs-melcas** | **fxo-melcas** | **e&m-melcas**}]

no tdm-group *tdm-group-no* **timeslot** *timeslot-list* [**type** {**e&m** | **fxs** [**loop-start** | **ground-start**] | **fxo** [**loop-start** | **ground-start**] | **fxs-melcas** | **fxo-melcas** | **e&m-melcas**}]

Syntax Description

tdm-group-no	TDM group number.
timeslot	Time-slot number.
timeslot-list	Time-slot list. The valid range is from 1 to 24 for T1, and from 1 to 15 and 17 to 31 for E1.
type	(Optional) (Valid only when the **mode cas** command is enabled.) Specifies the voice signaling type of the voice port. If configuring a TDM group for data traffic only, do not specify the type keyword. Choose from one of the following options: • **e&m**—for E&M signaling • **fxs**—for Foreign Exchange Station signaling (optionally, you can also specify loop-start or ground-start) • **fxo**—for Foreign Exchange Office signaling (optionally, you can also specify loop-start or ground-start) • **fxs-melcas**—for Foreign Exchange Station MEL CAS • **fxo-melcas**—for Foreign Exchange Office MEL CAS • **e&m-melcas**—for E&M Mercury Exchange Limited Channel-Associated signaling (MEL CAS) The MELCAS options apply only to E1 lines and are used primarily in the United Kingdom.

Defaults

No TDM group is configured.

Command Modes

Controller configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on Cisco MC38310 multiservice concentrators.

Release	Modification
12.1(1)T	This command was modified to include voice WAN interface cards (VWICs) for Cisco 2600 and Cisco 3600 series routers.
12.1(2)T	This command was modified for the OC-3/STM-1 ATM Circuit Emulation Service network module on the Cisco 2600 and 3600 series routers.

Usage Guidelines

The **tdm-group** command allows specific timeslots to switch from port 0 to port 1 and vice versa. This command is similar to the **channel-group** command, but it does not create a serial interface to terminate the specified channels.

Note Channel groups, CAS voice groups, and TDM groups all use group numbers. All group numbers configured for channel groups, CAS voice groups, and TDM groups must be unique on the local router. For example, you cannot use the same group number for a channel group and for a TDM group.

Examples

The following example shows TDM group 1 being set up to include timeslots 13 through 20:

```
controller T1 1
 tdm-group 1 timeslots 13-20
```

The following example configures TDM group number 20 on controller T1 1 to support FXO ground-start:

```
controller T1 1
 tdm-group 20 timeslot 20 type fxs ground-start
```

Related Commands

Command	Description
connect	Starts passage of data between ports for cross-connect TDM.

tech-prefix

To specify that a particular technology prefix be prepended to the destination pattern of a specific dial peer, use the **tech-prefix** command in dial-peer configuration mode. To disable the defined technology prefix for this dial peer, use the **no** form of this command.

tech-prefix *number*

no tech-prefix *number*

Syntax Description

number	Defines the numbers used as the technology prefix. Each technology prefix can contain up to 11 characters. Although not strictly necessary, a pound (#) symbol is frequently used as the last character in a technology prefix. Valid characters are 0 though 9, the pound (#) symbol, and the asterisk (*).

Defaults

No technology prefix is defined.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced on Cisco 2500 and 3600 series routers.

Usage Guidelines

Technology prefixes are used to distinguish between gateways that have specific capabilities within a given zone. In the exchange between the gateway and the gatekeeper, the technology prefix is used to select a gateway after the zone has been selected. Use the **tech-prefix** command to define technology prefixes.

Technology prefixes can be used as a discriminator so that the gateway can tell the gatekeeper that a certain technology is associated with a particular call (for example, 15# could mean a fax transmission), or a technology prefix can be used like an area code for more generic routing. No standard defines what the numbers in a technology prefix mean; by convention, technology prefixes are designated by a pound (#) symbol as the last character.

In most cases, there is a dynamic protocol exchange between the gateway and the gatekeeper that enables the gateway to inform the gatekeeper about technology prefixes and where to forward calls. If, for some reason, that dynamic registry feature is not in effect, you can statically configure the gatekeeper to query the gateway for this information by configuring the **gw-type-prefix** command on the gatekeeper. Use the **show gatekeeper gw-type-prefix** command to display how the gatekeeper has mapped the technology prefixes to local gateways.

Note Cisco gatekeepers use the asterisk (*) as a reserved character. If you are using Cisco gatekeepers, do not use the asterisk as part of the technology prefix.

Examples

The following example defines a technology prefix of 14# for the specified dial peer. In this example, the technology prefix means that the H.323 gateway will ask the RAS gatekeeper to direct calls using the technology prefix of 14#.

```
dial-peer voice 10 voip
 destination-pattern 14...
 tech-prefix 14#
```

Related Commands

Command	Description
gw-type-prefix	Configures a technology prefix in the gatekeeper.
show gatekeeper gw-type-prefix	Displays the gateway technology prefix table.

test call fallback probe

To test a probe to a particular IP address and display the Calculated Planning Impairment Factor (ICPIF) response time reporter (RTR) values, use the **test call fallback probe** command in EXEC mode. This command has no impact on the cache.

test call fallback probe *ip-address* [**codec** *711*/*729*]

Syntax Description

ip-address	Specifies the target IP address.
codec *711*/*729*	(Optional) Specifies a specific codec type.

Defaults

This command is not configured by default.

Command Modes

EXEC

Command History

Release	Modification
12.1(3)T	This command was introduced.

Examples

The following example demonstrates a test probe to IP address 10.1.1.4, and shows that the ICPIF value to 10.1.1.4 is 0:

```
Router# test call fallback probe 10.1.1.4

Running a test RTR probe....
ICPIF value for the test probe is 0
```

Related Commands

Command	Description
call fallback active	Enables fallback to alternate dial peers in case of network congestion.
call fallback monitor	Enables the monitoring of destinations without fallback to alternate dial peers.

test pots dial

To dial a telephone number for the plain old telephone service (POTS) port on the router by using a dial application on your workstation, use the **test pots dial** command in EXEC mode.

test pots *port* **dial** *number***[#]**

Syntax Description

port	Port number 1 or 2.
number	Telephone number to dial.
#	(Optional) Turns off dual tone multifrequency (DTMF) detection from the telephone while sending the *enbloc* signal. If you do not include the pound sign character (**#**) to terminate the *number* variable, you can use the telephone keypad to complete the call.

Command Modes

EXEC

Command History

Release	Modification
12.1(2)XF	The command **test pots** *port* **dial** was introduced on the Cisco 800 series routers.

Usage Guidelines

If the telephone is on the hook when you issue the dial command, the router rings the telephone, waits until the telephone is taken off the hook, and then dials the requested number. If the telephone is off the hook and providing a dial tone when you issue the command, the router dials the requested number.

Examples

The following POTS dial command dials the telephone number 4085551234:

```
Router# test pots 1 dial 4085551234#
```

For an example of the **test pots** *port* **dial** command with debug output, see the **debug pots csm** command in the *Cisco IOS Debug Command Reference*, Release 12.2.

Related Commands

Command	Description
show pots csm	Displays the current state of calls and the most recent event received by the CSM on the router.
test pots disconnect	Disconnects a telephone call for the POTS port on the router.

test pots disconnect

To disconnect a telephone call for the POTS port on the router, use the **test pots disconnect** command in EXEC mode.

test pots *port* **disconnect**

Syntax Description

port	Port number 1 or 2.

Command Modes

EXEC

Command History

Release	Modification
12.1(2)XF	This command was introduced on the Cisco 800 series routers.

Examples

The following POTS disconnect command disconnects a telephone call from POTS port 1:

```
Router# test pots 1 disconnect
```

For an example of the **test pots** *port* **disconnect** command with debug output, see the **debug pots csm** command in the *Cisco IOS Debug Command Reference*, Release 12.2.

Related Commands

Command	Description
show pots csm	Displays the current state of calls and the most recent event received by the CSM on the router.
test pots dial	Dials a telephone number for the POTS port on the router by using a dial application on your workstation.

test translation-rule

To test the execution of the translation rules on a specific name tag, use the **test translation-rule** command in global configuration mode. To disable the test, use the **no** form of this command.

test translation-rule *name-tag input-number* [*input-numbering-type*]

no test translation-rule *name-tag input-number* [*input-numbering-type*]

Syntax Description

name-tag	The tag number by which the rule set will be referenced. This is an arbitrarily chosen number. The range is 1 through 2,147,483,647.
input-number	The input string of digits for which a pattern matching is performed.
input-numbering-type	(Optional) The keyword choices for this field are **international**, **national**, **subscriber**, **abbreviated**, **unknown**, and **any**.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for Voice over IP on the Cisco AS5300.
12.0(7)XK	This command was first supported for the following voice technologies on the following platforms: • Voice over IP (Cisco 2600 series, Cisco 3600 series, Cisco MC3810 series) • Voice over Frame Relay (Cisco 2600 series, Cisco 3600 series, Cisco MC3810 multiservice concentrator) • Voice over ATM (Cisco 3600 series, Cisco MC3810 multiservice concentrator)
12.1(1)T	This command was first supported on the T train for the following voice technology on the following platforms: • Voice over IP (Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco 7200 series, and Cisco 7500 series)
12.1(2)T	This command was first supported on the T train for the following voice technologies on the following platforms: • Voice over IP (Cisco MC3810 multiservice concentrator) • Voice over Frame Relay (Cisco 2600 series, Cisco 3600 series, Cisco MC3810 multiservice concentrator) • Voice over ATM (Cisco 3600 series, Cisco MC3810 multiservice concentrator)

Examples

The following example shows output from the **test translation-rule** command:

```
Router# translation-rule 21
 Rule 1 555.% 1408555 subscriber international
 Rule 2 8.% 1408555 abbreviated international

Router# test translation-rule 21 45678 abbreviated
*Jan 19 16:39:14.578:The replace number 45614085558
```

Related Commands

Command	Description
numbering-type	Specifies number type for the VoIP or POTS dial peer.
rule	Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls.
show translation-rule	Displays the contents of all the rules that have been configured for a specific translation name.
translate	Applies a translation rule to a calling party number or a called party number for incoming calls.
translate-outgoing	Applies a translation rule to a calling party number or a called party number for outgoing calls.
translation-rule	Creates a translation name and enters translation-rule configuration mode.
voip-incoming translation-rule	Captures calls that originate from H.323-compatible clients.

test voice port detector

To test detector-related functions on a voice port, use the **test voice port detector** command in privileged EXEC mode.

Cisco 2600 and 3600 Series with Analog Voice Ports

test voice port *slot/subunit/port* **detector** {**m-lead** | **battery-reversal** | **ring** | **tip-ground** | **ring-ground** | **ring-trip**} {**on** | **off** | **disable**}

Cisco 2600 and 3600 Series with Digital Voice Ports

test voice port *slot/port***:***ds0-group* **detector** {**m-lead** | **battery-reversal** | **ring** | **tip-ground** | **ring-ground** | **ring-trip**} {**on** | **off** | **disable**}

Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

test voice port *slot/port* **detector** {**m-lead** | **battery-reversal** | **ring** | **tip-ground** | **ring-ground** | **ring-trip**} {**on** | **off** | **disable**}

Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

test voice port *slot***:***ds0-group* **detector** {**m-lead** | **battery-reversal** | **ring** | **tip-ground** | **ring-ground** | **ring-trip**} {**on** | **off** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 Series Routers with Analog Voice Ports:

slot/subunit/port	Tests the voice port that you specify with the *slot*/*subunit*/*port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1.

For the Cisco 2600 and 3600 Series Routers with Digital Voice Ports:

slot/*port***:***ds0-group*	Tests the voice port that you specify with the *slot*/*port***:***ds0-group* designation. • *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. • *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:

slot/port	Tests the voice port that you specify with the *slot/port* designation. • *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. • *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:

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

For All Platforms:

m-lead	Forces the E&M m-lead detector to the specified state.
loop	Forces the FXO loop detector to the specified state.
battery-reversal	Forces the FXO battery-reversal detector to the specified state.
ring	Forces the FXO ringing detector to the specified state.
tip-ground	Forces the FXO tip-ground detector to the specified state.
ring-ground	Forces the FXS ring-ground detector to the specified state.
ring-trip	Forces the FXS ring-trip detector to the specified state.
on	Forces the selected item to the on state.
off	Forces the selected item to the off state.
disable	Ends the forced state for the selected item.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Use the **test voice port detector** privileged EXEC command to force a detector into specific states for testing. For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. When you are finished testing, be sure to enter the command with the **disable** keyword to end the forced state. The **disable** keyword is available only if a test condition is already activated.

Examples

The following example forces the tip-ground detector to the off state on an FXO voice port (1/3) on a Cisco MC3810 and ends any call in progress:

```
Router# test voice port 1/3 detector tip-ground off
```

The following example ends the forced off state on an FXO voice port (1/3) on a Cisco MC3810:

```
Router# test voice port 1/3 detector tip-ground disable
```

The following example forces the ring-trip detector to the on state on an FXS port (0/0/1) on a Cisco 3600 series router and should start a call:

```
Router# test voice port 0/0/1 detector ring-trip on
```

The following example ends the forced on state on an FXS port (0/0/1) on a Cisco 3600 series router:

```
Router# test voice port 0/0/1 detector ring-trip disable
```

Related Commands

Command	Description
test voice port inject-tone	Injects a test tone into a voice port.
test voice port loopback	Performs loopback testing on a voice port.
test voice port relay	Tests relay-related functions on a voice port.
test voice port switch	Forces a voice port into fax or voice mode.

test voice port inject-tone

To inject a test tone into a voice port, use the **test voice port inject-tone** command in privileged EXEC mode.

Cisco 2600 and 3600 Series with Analog Voice Ports

test voice port *slot***/***subunit***/***port* **inject-tone** {**local** | **network**} {**1000hz** | **2000hz** | **200hz** | **3000hz** | **300hz** | **3200hz** | **3400hz** | **500hz** | **quiet** | **disable**}

Cisco 2600 and 3600 Series with Digital Voice Ports

test voice port *slot***/***port***:***ds0-group* **inject-tone** {**local** | **network**} {**1000hz** | **2000hz** | **200hz** | **3000hz** | **300hz** | **3200hz** | **3400hz** | **500hz** | **quiet** | **disable**}

Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

test voice port *slot***/***port* **inject-tone** {**local** | **network**} {**1000hz** | **2000hz** | **200hz** | **3000hz** | **300hz** | **3200hz** | **3400hz** | **500hz** | **quiet** | **disable**}

Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

test voice port *slot***:***ds0-group* **inject-tone** {**local** | **network**} {**1000hz** | **2000hz** | **200hz** | **3000hz** | **300hz** | **3200hz** | **3400hz** | **500hz** | **quiet** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 Series with Analog Voice Ports:

slot/subunit/port	Tests the voice port that you specify with the *slot*/*subunit*/*port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1.

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

*slot/port***:***ds0-group*	Tests the voice port that you specify with the *slot*/*port***:***ds0-group* designation. • *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. • *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:

slot/*port*	Tests the voice port that you specify with the *slot*/*port* designation. • *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. • *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:

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

For All platforms:

local	Directs the injected tone toward the local interface (near end).
network	Directs the injected tone toward the network (far end).
1000hz	Injects a 1-kilohertz test tone.
2000hz	Injects a 2-kilohertz test tone.
200hz	Injects a 200-hertz test tone.
3000hz	Injects a 3-kilohertz test tone.
300hz	Injects a 300-hertz test tone.
3200hz	Injects a 3.2-kilohertz test tone.
3400hz	Injects a 3.4-kilohertz test tone.
500hz	Injects a 500-hertz test tone.
quiet	Injects a quiet tone.
disable	Ends the test tone.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Use the **test voice port inject-tone** privileged EXEC command to inject a test tone or to end a test tone. A call must be established on the voice port under test. When you are finished testing, be sure to enter the **disable** keyword to end the test tone. The **disable** keyword is available only if a test condition is already activated.

When you enter the **disable** keyword, you must enter a direction (either **network** or **local**); however, you can enter either direction, regardless of which direction you entered to inject the test tone.

Examples

The following example injects a 1-kilohertz test tone into voice port 1/1, directed toward the network (far end), on a Cisco MC3810:

```
Router# test voice port 1/1 inject-tone network 1000hz
```

The following example removes the test tone from port 0/0/1 on a Cisco 3600 series router:

```
Router# test voice port 0/0/1 inject-tone network disable
```

or

```
Router# test voice port 0/0/1 inject-tone local disable
```

Related Commands

Command	Description
test voice port detector	Tests detector-related functions on a voice port.
test voice port loopback	Performs loopback testing on a voice port.
test voice port relay	Tests relay-related functions on a voice port.
test voice port switch	Forces a voice port into fax or voice mode.

test voice port loopback

To perform loopback testing on a voice port, use the **test voice port loopback** command in privileged EXEC mode.

Cisco 2600 and 3600 Series with Analog Voice Ports

test voice port *slot*/*subunit*/*port* **loopback** {**local** | **network** | **disable**}

Cisco 2600 and 3600 Series with Digital Voice Ports

test voice port *slot*/*port*:*ds0-group* **loopback** {**local** | **network** | **disable**}

Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

test voice port *slot*/*port* **loopback** {**local** | **network** | **disable**}

Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

test voice port *slot*:*ds0-group* **loopback** {**local** | **network** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 Series with Analog Voice Ports:

slot/*subunit*/*port*	Tests the voice port that you specify with the *slot*/*subunit*/*port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1.

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

slot/*port*:*ds0-group*	Tests the voice port that you specify with the *slot*/*port*:*ds0-group* designation. • *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. • *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:

slot/*port*	Tests the voice port that you specify with the *slot*/*port* designation. • *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. • *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:

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

All Platforms:

local	Forces a loopback at the voice port toward the customer premises equipment (CPE).
network	Forces a loopback at the voice port toward network.
disable	Ends the forced loopback.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Use the **test voice port loopback** privileged EXEC command to initiate or end a loopback at a voice port. A call must be established on the voice port under test. When you are finished testing, be sure to enter the **disable** keyword to end the forced loopback. The **disable** keyword is available only if a test condition is already activated.

Examples

The following example forces a loopback toward the CPE on voice port 1/1 on a Cisco MC3810:

```
Router# test voice port 1/1 loopback local
```

The following example ends a forced loopback on port 0/0/1 on a Cisco 3600 series router:

```
Router# test voice port 0/0/1 loopback disable
```

Related Commands

Command	Description
test voice port detector	Tests detector-related functions on a voice port.
test voice port inject-tone	Injects a test tone into a voice port.
test voice port relay	Tests relay-related functions on a voice port.
test voice port switch	Forces a voice port into fax or voice mode.

test voice port relay

To test relay-related functions on a voice port, use the **test voice port relay** command in privileged EXEC mode.

Cisco 2600 and 3600 Series with Analog Voice Ports

test voice port *slot/subunit/port* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off** | **disable**}

Cisco 2600 and 3600 Series with Digital Voice Ports

test voice port *slot/port*:*ds0-group* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off** | **disable**}

Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

test voice port *slot/port* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off** | **disable**}

Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

test voice port *slot*:*ds0-group* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 Series with Analog Voice Ports:

slot/subunit/port	Tests the voice port that you specify with the *slot/subunit/port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1.

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

slot/port:*ds0-group*	Tests the voice port that you specify with the *slot/port*:*ds0-group* designation. • *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. • *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:

slot/*port*	Tests the voice port that you specify with the *slot*/*port* designation. • *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. • *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:

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

All Platforms:

e-lead	Forces the E&M e-lead relay to the specified state.
loop	Forces the FXO loop relay to the specified state.
ring-ground	Forces the FXO ring-ground relay to the specified state.
battery-reversal	Forces the FXO battery-reversal relay to the specified state.
power-denial	Forces the FXS power-denial relay to the specified state.
ring	Forces the FXS ringing relay to the specified state.
tip-ground	Forces the FXS tip-ground relay to the specified state.
on	Forces the selected item to the on state.
off	Forces the selected item to the off state.
disable	Ends the forced state for the selected item.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Use the **test voice port relay** privileged EXEC command to force a relay into specific states for testing. For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. When you are finished testing, be sure to enter the **disable** keyword to end the forced state. The **disable** keyword is available only if a test condition is already activated.

Examples

The following example forces the E&M e-lead relay to the on state on port 0/0/1 on a Cisco 3600 series router:

```
Router# test voice port 0/0/1 relay e-lead on
```

The following example ends a forced actuation of the battery-reversal relay on an FXS port (0/0/1) on a Cisco 3600 series router:

```
Router# test voice port 0/0/1 relay battery-reversal disable
```

Related Commands

Command	Description
test voice port detector	Tests detector-related functions on a voice port.
test voice port inject-tone	Injects a test tone into a voice port.
test voice port switch	Forces a voice port into fax or voice mode.

test voice port switch

To force a voice port into fax mode, use the **test voice port switch** command in privileged EXEC mode.

Cisco 2600 and 3600 Series with Analog Voice Ports

test voice port *slot***/***subunit***/***port* **switch** {**fax** | **disable**}

Cisco 2600 and 3600 Series with Digital Voice Ports

test voice port *slot***/***port***:***ds0-group* **switch** {**fax** | **disable**}

Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

test voice port *slot***/***port* **switch** {**fax** | **disable**}

Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

test voice port *slot***:***ds0-group* **switch** {**fax** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 Series with Analog Voice Ports:

slot/subunit/port	Tests the voice port that you specify with the *slot/subunit/port* designation. • *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *subunit* specifies a voice interface card (VIC) in which the voice port is located. Valid entries are 0 and 1. • *port* specifies an analog voice port number. Valid entries are 0 and 1.

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

slot/port:ds0-group	Tests the voice port that you specify with the *slot/port:ds0-group* designation. • *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. • *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. • *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:

slot/port	Tests the voice port that you specify with the *slot/port* designation. • *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. • *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:

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

For All Platforms:

fax	Forces a switch to fax mode.
disable	Ends fax mode; switches back to voice mode.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Use the **test voice port switch** privileged EXEC command to force a voice port into fax mode for testing. If no fax data is detected by the voice port, the voice port remains in fax mode for 30 seconds and then reverts automatically to voice mode. After you enter the **test voice port switch fax** command, you can use the **show voice call** or **show voice call summary** command to check whether the voice port is able to operate in fax mode.

The **disable** keyword ends the forced mode switch; however, the fax mode ends automatically after 30 seconds. The **disable** keyword is available only while the voice port is in fax mode.

Examples

The following example forces voice port 1/3 on a Cisco MC3810 into fax mode:

```
Router# test voice port 1/3 switch fax
```

The following example returns voice port 0/0/1 on a Cisco 3600 series router to voice mode:

```
Router# test voice port 0/0/1 switch disable
```

Related Commands

Command	Description
show voice call	Displays the call processing and protocol state-machine information for a voice port.
show voice call summary	Displays a summary of the call processing and protocol state-machine information for a voice port.

test vrm busyout

To busy out a specific digital signal processor (DSP) or channels on a specific DSP, use the **test vrm busyout** command in privileged EXEC mode.

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

Syntax Description

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

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(7)T	This command was introduced on Cisco AS5800 universal access servers.

Usage Guidelines

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

Examples

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

```
Router# test vrm busyout 4 all
```

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

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

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

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

Related Commands

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

test vrm reset

To reset a particular digital signal processor (DSP), use the **test vrm reset** command in privileged EXEC mode.

test vrm reset *slot-number dsp-number*

Syntax Description

slot-number	Number that identifies the slot in which the voice feature card (VFC) is installed.
dsp-number	Number that identifies the DSP to be reset.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco AS5300.

Usage Guidelines

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

Examples

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

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

test vrm unbusyout

To restore activity to a busied-out digital signal processor (DSP) or busied-out channels on a DSP, use the **test vrm unbusyout** command in privileged EXEC mode.

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

Syntax Description

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

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(7)T	This command was introduced on the Cisco AS5300.

Usage Guidelines

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

Examples

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

```
Router# test vrm unbusyout 4 all
```

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

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

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

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

Related Commands

Command	Description
test vrm busyout	Busy outs a specific DSP or channels on a specific DSP.

threshold noise

To configure a noise threshold for incoming calls, use the **threshold noise** command in voice-port configuration mode. To restore the default, use the **no** form of this command.

threshold noise {*value*}

no threshold noise {*value*}

Syntax Description

value	Number that establishes a noise threshold. Valid values are from -30 to -90 decibels (dBs). The default value is -62 dB.

Defaults

-62dB

Command Modes

Voice-port configuration

Command History

Release	Modification
12.2(16)	This command was introduced on the following platforms: Cisco 1700 Cisco 1751, Cisco 2600 (with and without the NM-HDA), Cisco 3600 (with and without the NM-HDA), Cisco 7200 (with and without the NM-HDA), Cisco AS5300, Cisco AS5800, and Cisco MC3810.

Usage Guidelines

Cisco voice activity detection (VAD) has two layers: application programming interface (API) layer and processing layer. There are 3 states that the processing layer classifies incoming signals: speech, unknown, and silence. The state of the incoming signals is determined by the noise threshold.

In earlier Cisco IOS Releases, the noise threshold is fixed between -62dB and -78 dB. If the voice level is below the noise threshold, then the signal is classified as silence. If the incoming signal cannot be classified, the variable thresholds that are computed with the statistics of speech and noise that VAD gathers is used to make a determination. If the signal still cannot be classified, then it is marked as unknown. The final decision is made by the API. For applications such as hoot-n-holler, you could have the noise create unwanted spurious packets (for example, a voice stream) taking up bandwidth.

With Cisco IOS Release 12.2(16), the noise threshold is configurable using the **threshold noise** command.

Examples

The following sample configuration shows a noise threshold level of -50 dB configured on a Cisco 3600:

```
voice-port 1/0/0
 threshold noise -50
```

timeouts call-disconnect

To configure the call disconnect timeout value for a specified voice port, use the **timeouts call-disconnect** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

timeouts call-disconnect *seconds*

no timeouts call-disconnect

Syntax Description

seconds	Sets the call-disconnect timeout duration, in seconds. Valid values are from 0 to 120.

Defaults

60 seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(9)T	This command was introduced on Cisco 3600 series routers.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.

Usage Guidelines

This command applies to Cisco 3600 series routers. To disable the timeouts call-disconnect timer, set the seconds value to 0. Use the **timeouts call-disconnect** command to specify the number of seconds for which the originating end system waits after receiving disconnect before notifying the user to hang up by playing a fast busy tone. During this duration, the user just hears silence. If the command is disabled by setting the value to 0, the user hears silence indefinitely.

Examples

The following example sets a call-disconnect timeout value of 10 seconds on a Cisco 3600 series router voice port:

```
voice-port 1/0/0
 timeouts call-disconnect 10
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timing delay-duration	Configures the delay dial signal duration for a specified voice port.

timeouts initial

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

Defaults

10 seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.

Usage Guidelines

Use the **timeouts initial** command to specify the number of seconds for which the system will wait for the caller to input the first digit of the dialed digits. The timeouts initial timer is activated when the call is accepted and is deactivated when the caller inputs the first digit. If the configured timeout value is exceeded, the caller is notified through the appropriate tone and the call is terminated.

To disable the timeouts initial timer, set the *seconds* value to 0.

Examples

The following example sets the initial digit timeout value on the Cisco 3600 series to 10 seconds:

```
voice-port 1/0/0
 timeouts initial 10
```

The following example sets the initial digit timeout value on the Cisco MC3810 multiservice concentrator to 10 seconds:

```
voice-port 1/1
 timeouts initial 10
```

Related Commands

Command	Description
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.

timeouts interdigit

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

Defaults

10 seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.

Usage Guidelines

This command applies to both the Cisco 3600 series router and the Cisco MC3810 multiservice concentrator.

Use the **timeouts interdigit** command to specify the number of seconds for which the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The timeouts interdigit timer is activated when the caller inputs a digit and is restarted each time the caller inputs another digit until the destination address is identified. If the configured timeout value is exceeded before the destination address is identified, the caller is notified through the appropriate tone and the call is terminated.

To disable the timeouts interdigit timer, set the *seconds* value to 0.

Examples

The following example sets the interdigit timeout value on the Cisco 3600 series for 10 seconds:

```
voice-port 1/0/0
 timeouts interdigit 10
```

The following example sets the interdigit timeout value on the Cisco MC3810 multiservice concentrator for 10 seconds:

```
voice-port 1/1
 timeouts interdigit 10
```

Related Commands

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

timeouts ringing

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

timeouts ringing {*seconds* | **infinity**}

no timeouts ringing

Syntax Description

seconds	The duration, in seconds, for which a voice port allows ringing to continue if a call is not answered. The range is from 5 to 60,000. The default is 180.
infinity	Ringing continues until the caller goes on-hook.

Defaults

180 seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators.
12.1(2)T	This command was integrated into Cisco Release 12.1(2)T.

Usage Guidelines

The **timeouts ringing** command provides the capability to limit the length of time for which a caller can continue ringing a telephone when there is no answer.

Examples

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

```
voice-port 1/1
 timeouts ringing 600
```

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

```
voice-port 0/0/1
 timeouts ringing 600
```

Related Commands

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

timeouts wait-release

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

timeouts wait-release {*seconds* | **infinity**}

no timeouts wait-release

Syntax Description

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

Defaults

30 seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1) MA	This command was introduced on the Cisco MC3810 series.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

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

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

Examples

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

```
voice-port 1/1
 timeouts wait-release 180
```

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

```
voice-port 0/0/1
 timeouts wait-release 180
```

Related Commands

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

timers

To configure the Session Initiation Protocol (SIP) signaling timers, use the **timers** command in the Session Initiation Protocol (SIP) user agent configuration mode. To restore the default value, use the **no** form of this command.

timers {**trying** *number* | **connect** *number* | **disconnect** *number* | **expires** *number*}

no timers {**trying** *number* | **connect** *number* | **disconnect** *number* | **expires** *number*}

Syntax Description

trying *number*	Time (in milliseconds) to wait for a 100 response to an INVITE request. Possible values are 100 through 1000. The default is 500.
connect *number*	Time (in milliseconds) to wait for a 200 response to an ACK request. Possible values are 100 through 1000. The default is 500.
disconnect *number*	Time (in milliseconds) to wait for a 200 response to a BYE request. Possible values are 100 through 1000. The default is 500.
expires *number*	Time (in milliseconds) for which an INVITE request is valid. Possible values are 60,000 through 300,000. The default is 180,000.

Defaults

The default for trying, connect, and disconnect is 500. The default for expires is 180,000.

Command Modes

SIP user agent configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on Cisco 2600 and 3600 series routers and Cisco AS5300 universal access servers.
12.1(3)T	This command was modified to change the names of the parameters. Two of the parameters (invite-wait-180 and invite-wait-200) were combined into one (trying).

Usage Guidelines

If you used the previous version of this command to configure timers, your previous timer settings will be maintained. The output of the **show running configuration** command will reflect both timers.

To reset this command to the default value, you can also use the **default** command.

Examples

The following example configures the SIP signaling timers to wait 500 milliseconds for a 100 response to an INVITE request:

```
sip-ua
 timers trying 500
```

timing clear-wait

To indicate the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port, use the **timing clear-wait** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing clear-wait *milliseconds*

no timing clear-wait *milliseconds*

Syntax Description

milliseconds	Minimum amount of time, in milliseconds, between the inactive seizure signal and the call being cleared. Valid entries on the Cisco 3600 series are numbers from 200 to 2000. Valid entries on the Cisco MC3810 are numbers from 100 to 2000. Supported on E&M ports only.

Defaults

400 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 2600 and 3600 series routers.

Examples

The following example configures the clear-wait duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing clear-wait 300
```

The following example configures the clear-wait duration on a Cisco MC3810 multiservice concentrator voice port to 300 milliseconds:

```
voice-port 1/1
 timing clear-wait 300
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing delay-duration

To specify the delay signal duration for a specified voice port, use the **timing delay-duration** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing delay-duration *milliseconds*

no timing delay-duration *milliseconds*

Syntax Description

milliseconds	Delay signal duration for delay dial signaling, in milliseconds. Valid entries are numbers from 100 to 5000. Supported on E&M ports only.

Defaults

2000 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.

Usage Guidelines

The call direction for the **timing delay-duration** command is out.

Examples

The following example configures the delay signal duration on a Cisco 3600 series voice port for 3000 milliseconds:

```
voice-port 1/0/0
 timing delay-duration 3000
```

The following example configures the delay signal duration on a Cisco MC3810 multiservice concentrator voice port for 3000 milliseconds:

```
voice-port 1/1
 timing delay-duration 3000
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing delay-start

To specify the minimum delay time from outgoing seizure to out-dial address for a specified voice port, use the **timing delay-start** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing delay-start *milliseconds*

no timing delay-start *milliseconds*

Syntax Description

milliseconds	Minimum delay time, in milliseconds, from outgoing seizure to outdial address. Valid entries are numbers from 20 to 2000. Supported on E&M ports only.

Defaults

300 milliseconds on the Cisco 3600 series.

150 milliseconds on the Cisco MC3810 multiservice concentrator.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.

Usage Guidelines

The call direction for the **timing delay-start** command is out.

Examples

The following example configures the delay-start duration on a Cisco 3600 series voice port for 250 milliseconds:

```
voice-port 1/0/0
 timing delay-start 250
```

The following example configures the delay-start duration on a Cisco MC3810 multiservice concentrator voice port for 250 milliseconds:

```
voice-port 1/1
 timing delay-start 250
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing delay-with-integrity

To specify the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810, use the **timing delay-with-integrity** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing delay-with-integrity *milliseconds*

no timing delay-with-integrity *milliseconds*

Syntax Description

milliseconds	Duration of the wink pulse for the delay dial, in milliseconds. Valid entries are numbers from 0 to 5000. Supported on E&M ports only.

Defaults

Zero (0)

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on Cisco MC3810 multiservice concentrators.

Usage Guidelines

This command applies only to the Cisco MC3810 multiservice concentrator.

Examples

The following example configures the duration of the wink pulse for the delay dial on a Cisco MC3810 voice port for 10 milliseconds:

```
voice-port 1/1
 timing delay-with-integrity 10
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing dial-pulse min-delay

To specify the time between wink-like pulses for a specified voice port, use the **timing dial-pulse min-delay** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing dial-pulse min-delay *milliseconds*

no timing dial-pulse min-delay *milliseconds*

Syntax Description

milliseconds	Time, in milliseconds, between the generation of wink-like pulses. Valid entries are integers from 0 to 5000.

Defaults

300 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.

Usage Guidelines

Use the **timing dial-pulse min-delay** command with PBXs that require a wink-like pulse, even though they have been configured for delay-dial signaling. If the value for this argument is set to 0, the router will not generate this wink-like pulse. The call signal direction for this command is in.

Examples

The following example configures the time between the generation of wink-like pulses on a Cisco 3600 series voice port for 350 milliseconds:

```
voice-port 1/0/0
 timing dial-pulse min-delay 350
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing dialout-delay

To specify the dial-out delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator, use the **timing dialout-delay** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing dialout-delay *milliseconds*

no timing dialout-delay *milliseconds*

Syntax Description

milliseconds	Dial-out delay, in milliseconds, for the sending digit or cut-through on an FXO trunk or an E&M immediate trunk. Valid entries are from 100 to 5000 milliseconds.

Defaults

300 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on Cisco MC3810 multiservice concentrators.

Usage Guidelines

This command applies only to the Cisco MC3810 multiservice concentrator.

Examples

The following example configures the dial-out delay on a Cisco MC3810 multiservice concentrator voice port for 350 milliseconds:

```
voice-port 1/1
 timing dialout-delay 350
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing digit

To specify the dual tone multifrequency (DTMF) digit signal duration for a specified voice port, use the **timing digit** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing digit *milliseconds*

no timing digit *milliseconds*

Syntax Description

milliseconds	The DTMF digit signal duration, in milliseconds. Valid entries are integers from 50 to 100. Supported on FXO, FXS and E&M ports.

Defaults

100 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.

Usage Guidelines

The call signal direction for the **timing digit** command is out.

Examples

The following example configures the DTMF digit signal duration on a Cisco 3600 series voice port for 50 milliseconds:

```
voice-port 1/0/0
 timing digit 50
```

The following example configures the DTMF digit signal duration on a Cisco MC3810 multiservice concentrator voice port for 50 milliseconds:

```
voice-port 1/1
 timing digit 50
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing guard-out

To specify the guard-out duration of an Foreign Exchange Office (FXO) voice port, use the **timing guard-out** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

timing guard-out *milliseconds*

no timing guard-out

Syntax Description

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

Defaults

2000 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810 multiservice concentrator
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **timing guard-out** command applies to the Cisco 2600, 3600, and MC3810 multiservice concentrator. This command is supported on FXO voice ports only.

Examples

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

```
voice-port 1/1
 timing guard-out 1000
```

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

```
voice-port 1/0/0
 timing guard-out 1000
```

timing hookflash-input

To specify the maximum duration of a hookflash for an Foreign Exchange Station (FXS) interface, use the **timing hookflash-input** command in voice-port configuration mode. To restore the default duration for hookflash timing, use the **no** form of this command.

timing hookflash-input *milliseconds*

no timing hookflash-input

Syntax Description

milliseconds	Duration of the hookflash, in milliseconds. Possible values are 50 through 1550 milliseconds. Default is 600 milliseconds.

Defaults

600 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 3600 series.

Usage Guidelines

This command does *not* affect whether hookflash relay is enabled; hookflash relay is enabled only when the **dtmf-relay h245-signal** command is configured on the applicable VoIP dial peers. When the **dtmf-relay h245-signal** command is configured, the H.323 gateway relays hookflash by using an H.245 "signal" User Input Indication method.Hookflash is sent only when an h245 signal is available.

Use the **timing hookflash-input** command on FXS interfaces to specify the maximum duration (in milliseconds) of a hookflash indication. If the hookflash lasts longer than the specified limit, the FXS interface processes the indication as an on-hook.

Examples

The following example implements timing for the hookflash with a duration of 200 milliseconds:

```
voice-port 1/0/0
 timing hookflash-input 200
```

Related Commands

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

timing hookflash-out

To specify the duration of hookflash indications that the gateway generates on a Foreign Exchange Office (FXO) interface, use the **timing hookflash-out** command in voice-port configuration mode. To restore the default duration for hookflash timing, use the **no** form of this command.

timing hookflash-out *milliseconds*

no timing hookflash-out

Syntax Description

milliseconds	Duration of the hookflash, in milliseconds. Possible values are 50 through 1550 milliseconds.

Defaults

400 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on Cisco 2500, Cisco 2600, Cisco 3600, and Cisco 7200 series routers and Cisco MC3810 multiservice concentrators.

Usage Guidelines

This command does *not* affect whether hookflash relay is enabled; hookflash relay is enabled only when the **dtmf-relay h245-signal** command is configured on the applicable Voice over IP (VoIP) dial peers. Hookflash is relayed by using an h245-signal indication and can be sent only when an h245 signal is available.

Use the **timing hookflash-out** command on FXO interfaces to specify the duration (in milliseconds) of a hookflash indication. To set hookflash timing parameters for analog voice interfaces, use the voice-port **timing** subcommand.

Examples

The following example implements timing for the hookflash with a duration of 200 milliseconds:

```
Router# voice-port 1/0/0
 timing hookflash-out 200
```

Related Commands

Command	Description
dtmf-relay (Voice over IP)	Specifies how an H.323 gateway relays DTMF tones between telephony interfaces and an IP network.
voice-port	Enters voice-port configuration mode.

timing interdigit

To specify the dual-tone multifrequency (DTMF) interdigit duration for a specified voice port, use the **timing interdigit** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing interdigit *milliseconds*

no timing interdigit *milliseconds*

Syntax Description

milliseconds	DTMF interdigit duration, in milliseconds. Valid entries are numbers from 50 to 500 milliseconds. Supported on FXO, FXS and E&M ports.

Defaults

100 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.
11.3(1)MA	This command was supported on Cisco MC3810 multiservice concentrators.

Usage Guidelines

The call signal direction for the **timing interdigit** command is out.

Examples

The following example configures the DTMF interdigit duration on a Cisco 3600 series voice port for 150 milliseconds:

```
voice-port 1/0/0
 timing interdigit 150
```

The following example configures the DTMF interdigit duration on a Cisco MC3810 multiservice concentrator voice port for 150 milliseconds:

```
voice-port 1/1
 timing interdigit 150
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing percentbreak

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

timing percentbreak *percent*

no timing percentbreak

Syntax Description

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

Defaults

50 percent

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA4	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **timing percentbreak** command is supported on Foreign Exchange Office (FXO) and ear and mouth (E&M) voice ports only.

Examples

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

```
voice-port 1/1
 timing percentbreak 30
```

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

```
voice-port 0/0/1
 timing percentbreak 30
```

Related Commands

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

timing pulse

To specify the pulse dialing rate for a specified voice port, use the **timing pulse** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing pulse *pulses-per-second*

no timing pulse *pulses-per-second*

Syntax Description

pulses-per-second	Pulse dialing rate, in pulses per second. Valid entries are numbers from 10 to 20. Supported on Foreign Exchange Office (FXO) and ear and mouth (E&M) ports only.

Defaults

20 pulses per seconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.
11.3(1)MA	This command was supported on Cisco MC3810 multiservice concentrators.

Usage Guidelines

The call signal direction for the **timing pulse** command is out.

Examples

The following example configures the pulse dialing rate on a Cisco 3600 series voice port for 15 pulses per second:

```
voice-port 1/0/0
 timing pulse 15
```

The following example configures the pulse dialing rate on a Cisco MC3810 multiservice concentrator voice port for 15 pulses per second:

```
voice-port 1/1
 timing pulse 15
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.

Command	Description
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing pulse-interdigit

To specify the pulse interdigit timing for a specified voice port, use the **timing pulse-interdigit** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing pulse-interdigit *milliseconds*

no timing pulse-interdigit *milliseconds*

Syntax Description

milliseconds	Pulse dialing interdigit timing, in milliseconds. Valid entries are integers from 100 to 1000. Supported on Foreign Exchange Office (FXO) and ear and mouth (E&M) ports only.

Defaults

500 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.
11.3(1)MA	This command was supported on Cisco MC3810 multiservice concentrators.

Usage Guidelines

The call signal direction for the **timing pulse-interdigit** command is out.

Examples

The following example configures the pulse-dialing interdigit timing on a Cisco 3600 series voice port for 300 milliseconds:

```
voice-port 1/0/0
 timing pulse-interdigit 300
```

The following example configures the pulse-dialing interdigit timing on a Cisco MC3810 multiservice concentrator voice port for 300 milliseconds:

```
voice-port 1/1
 timing pulse-interdigit 300
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing wink-duration

To specify the maximum wink-signal duration for a specified voice port, use the **timing wink-duration** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

timing wink-duration *milliseconds*

no timing wink-duration *milliseconds*

Syntax Description

milliseconds	Maximum wink-signal duration, in milliseconds, for a wink-start signal. Valid entries are from 100 to 400 milliseconds. Supported on ear and mouth (E&M) ports only.

Defaults

200 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.
11.3(1)MA	This command was supported on Cisco MC3810 multiservice concentrators.

Usage Guidelines

The call signal direction for the **timing wink-duration** command is out.

Examples

The following example configures the wink-signal duration on a Cisco 3600 series voice port for 300 milliseconds:

```
voice-port 1/0/0
 timing wink-duration 300
```

The following example configures the wink-signal duration on a Cisco MC3810 multiservice concentrator voice port for 300 milliseconds:

```
voice-port 1/1
 timing wink-duration 300
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing delay-with-integrity	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing wink-wait

To specify the maximum wink-wait duration for a specified voice port, use the **timing wink-wait** command in voice-port configuration mode. To reset the default value, use the **no** form of this command.

timing wink-wait *milliseconds*

no timing wink-wait *milliseconds*

Syntax Description

milliseconds	Maximum wink-wait duration, in milliseconds, for a wink start signal. Valid entries are from 100 to 5000 milliseconds. Supported on ear and mouth (E&M) ports only.

Defaults

200 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.
11.3(1)MA	This command was supported on Cisco MC3810 multiservice concentrators.

Usage Guidelines

The call signal direction for the **timing wink-wait** command is out.

Examples

The following example configures the wink-wait duration on a Cisco 3600 series voice port for 300 milliseconds:

```
voice-port 1/0/0
 timing wink-wait 300
```

The following example configures the wink-wait duration on a Cisco MC3810 multiservice concentrator voice port for 300 milliseconds:

```
voice-port 1/1
 timing wink-wait 300
```

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports on the Cisco MC3810 multiservice concentrator.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.

Command	Description
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810 multiservice concentrator.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.

token-root-name

To specify which root or Certificate Authority (CA) certificate the router should use to validate the settlement token in the incoming setup message, use the **token-root-name** command in settlement configuration mode. To restore the default value, use the **no** form of this command.

token-root-name *name*

no token-root-name *name*

Syntax Description

name	Specifies the name that is the certificate identification as configured through the **crypto ca identity** *name* command or the **crypto ca trusted-root** *name* command.

Defaults

The terminating gateway uses the CA certificate to validate the settlement token.

Command Modes

Settlement configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on Cisco 2600 and 3600 series routers and Cisco AS5300 and AS5800 universal access servers.

Examples

The following example defines the **token-root-name** as sample:

```
token-root-name sample
```

The following example shows new output for the **show settlement** command to display the value of the **token-root-name** command:

```
Settlement Provider 0
        Operation Status = UP
        Type = osp
        Address url = https://1.14.115.100:8444/
        Encryption = all                (default)
        Token Root Name = sample
        Max Concurrent Connections = 20 (default)
        Connection Timeout = 3600 (s)   (default)
        Response Timeout = 1 (s)        (default)
        Retry Delay = 2 (s)             (default)
        Retry Limit = 1                 (default)
        Session Timeout = 86400 (s)     (default)
        Customer Id = 1000
        Device Id = 2000
        Roaming = Disabled              (default)
        Signed Token = On

        Number of Connections = 1
        Number of Transactions = 0
```

Related Commands

Command	Description
crypto ca identity	Declares the Certificate Authority that your router should use.
crypto ca trusted-root	Configures the root certificate that the server uses to sign the settlement tokens.
show settlement	Displays the configuration for all settlement server transactions.

tone ringback alert-no-PI

To generate automatic ringback for the caller when no Progress Indicator (PI) alert has been received over the H.323 network, use the **tone ringback alert-no-PI** command in dial-peer configuration mode. To disable automatic ringback, use the **no** form of this command.

tone ringback alert-no-PI

no tone ringback alert-no-PI

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(1)	This command was introduced on Cisco 1700 series, Cisco 2600 series, Cisco 3600 series, Cisco 3700 series, and Cisco 7200 series routers and on the Cisco AS5300 and Cisco AS5800 universal access servers.

Usage Guidelines

The **tone ringback alert-no-PI** command is used to generate ringback in an H.323 network when the attached device (for example, an ISDN device) cannot.

Examples

The following example activates ringback for a VoIP dial peer numbered 322:

```
router(config)# dial-peer voice 322 voip
router(config-dial-peer)# tone ringback alert-no-PI
```

Related Commands

Command	Description
progress_ind	Sets a specific PI in call Setup, Progress, or Connect messages from an H.323 VoIP gateway.

translate

To apply a translation rule to an inbound plain old telephone service (POTS) call leg, use the **translate** command in voice-port configuration mode. To remove the translation rule to an inbound POTS call leg, use the **no** form of this command.

translate {**calling-number** | **called-number**} *name-tag*

no translate {**calling-number** | **called-number**} *name-tag*

Syntax Description

calling-number	Applies the translation rule to the inbound calling party number.
called-number	Applies the translation rule to the inbound called party number.
name-tag	The tag number by which the rule set will be referenced. This is an arbitrarily chosen number. The range is 1 through 2,147,483,647.

Defaults

No default behavior or values.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for Voice over IP on the Cisco AS5300.
12.0(7)XK	This command was first supported for Voice over IP on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 multiservice concentrator.
12.1(1)T	This command was first supported on the T train for Voice over IP on the Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco 7200 series, and Cisco 7500 series.
12.1(2)T	This command was first supported on the T train for Voice over IP on the Cisco MC3810 multiservice concentrator.

Examples

The following example applies translation rule 21 to the POTS inbound calling party number:

```
translation-rule 21
 rule 1 555.% 1408555 subscriber international
 rule 2 7.% 1408555 abbreviated international
voice-port 0:1
 translate calling-number 21
```

The following example applies translation rule 20 to the POTS inbound called party number:

```
translation-rule 20
 rule 1 .%555.% 7 any abbreviated
voice-port 0:1
 translate called-number 20
```

Related Commands

Command	Description
numbering-type	Specifies number type for the VoIP or POTS dial peer.
rule	Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls.
show translation-rule	Displays the contents of all the rules that have been configured for a specific translation name.
translate-outgoing	Applies a translation rule to a calling party number or a called party number for outgoing calls.
translation-rule	Creates a translation name and enters translation-rule configuration mode.
voip-incoming translation-rule	Captures calls that originate from H.323-compatible clients.

translate-outgoing

To apply a translation rule to an outbound plain old telephone service (POTS) or Voice over IP (VoIP) call leg, use the **translate-outgoing** command in dial-peer configuration mode. To remove the translation rule to an outbound POTS or VoIP call leg, use the **no** form of this command.

translate-outgoing {**calling-number** | **called-number**} *name-tag*

no translate-outgoing {**calling-number** | **called-number**} *name-tag*

Syntax Description

calling-number	Applies the translation rule to the outbound calling party number.
called-number	Applies the translation rule to the outbound called party number.
name-tag	The tag number by which the rule set will be referenced. This is an arbitrarily chosen number. The range is 1 through 2,147,483,647.

Defaults

No default behavior or values.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for Voice over IP on the Cisco AS5300.
12.0(7)XK	This command was first supported for Voice over IP on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 multiservice concentrator.
12.1(1)T	This command was first supported on the T train for Voice over IP on the Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco 7200 series, and Cisco 7500 series.
12.1(2)T	This command was first supported on the T train for Voice over IP on the Cisco MC3810 multiservice concentrator.

Examples

The following example applies translation rule 21 to the VoIP outbound calling number:

```
Translation-rule 21
 rule 1 555.% 1408555 subscriber international
 rule 2 7.% 1408555 abbreviated international
 dial-peer voice 100 voip
 translate-outgoing calling-number 21
```

The following example applies translation rule 20 to the VoIP called number:

```
translation-rule 20
 rule 1 .%555.% 7 any abbreviated
dial-peer voice 100 voip
 translate-outgoing called-number 20
```

Related Commands

Command	Description
numbering-type	Specifies number type for the VoIP or POTS dial peer.
rule	Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls.
show translation-rule	Displays the contents of all the rules that have been configured for a specific translation name.
translate	Applies a translation rule to a calling party number or a called party number for incoming calls.
translation-rule	Creates a translation name and enters translation-rule configuration mode.
voip-incoming translation-rule	Captures calls that originate from H.323-compatible clients.

translation-rule

To create a translation name and enter translation-rule configuration mode to apply rules to the translation name, use the **translation-rule** command in global configuration mode. To remove the translation rule, use the **no** form of this command.

translation-rule *name-tag*

no translation-rule *name-tag*

Syntax Description

name-tag	The tag number by which the rule set will be referenced. This is an arbitrarily chosen number. The range is 1 through 2,147,483,647.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for Voice over IP on the Cisco AS5300.
12.0(7)XK	This command was first supported for the following voice technologies on the following platforms: • Voice over IP (Cisco 2600 series, Cisco 3600 series, Cisco MC3810 multiservice concentrator) • Voice over Frame Relay (Cisco 2600 series, Cisco 3600 series, Cisco MC3810 multiservice concentrator) • Voice over ATM (Cisco 3600 series, Cisco MC3810 multiservice concentrator)
12.1(1)T	This command was first supported on the T train for the following voice technology on the following platforms: Voice over IP (Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco 7200 series, and Cisco 7500 series)
12.1(2)T	This command was first supported on the T train for the following voice technologies on the following platforms: • Voice over IP (Cisco MC3810 multiservice concentrator) • Voice over Frame Relay (Cisco 2600 series, Cisco 3600 series, Cisco MC3810 multiservice concentrator) • Voice over ATM (Cisco 3600 series, Cisco MC3810 multiservice concentrator)

Usage Guidelines

This command applies to all translation rules.

Examples

The following example creates translation rule 21 and applies a rule to it:

```
translation-rule 21
 rule 1 555.% 1408555 subscriber international
```

Related Commands

Command	Description
numbering-type	Specifies number type for the VoIP or POTS dial peer.
rule	Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls.
test translation-rule	Tests the execution of the translation rules on a specific name tag.
translate	Applies a translation rule to a calling party number or a called party number for incoming calls.
translate-outgoing	Applies a translation rule to a calling party number or a called party number for outgoing calls.
voip-incoming translation-rule	Captures calls that originate from H.323-compatible clients.

transport

To configure the Session Inititation Protocol (SIP) user agent (gateway) for SIP signaling messages on inbound calls through the SIP TCP or UDP socket, use the **transport** command in SIP user-agent configuration mode. To block reception of SIP signaling messages on a particular socket, use the **no** form of this command.

transport {**udp** | **tcp**}

no transport {**udp** | **tcp**}

Syntax Description

udp	Configures the SIP user agent to receive SIP messages on UDP port 5060.
tcp	Configures the SIP user agent to receive SIP messages on TCP port 5060.

Defaults

Both UDP and TCP transport protocols are enabled.

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on Cisco 2600 and 3600 series routers and Cisco AS5300 universal access servers.
12.1(3)T	Support for access platforms was added.

Usage Guidelines

This command controls whether messages reach the SIP service provider interface (SPI).

Examples

The following example configures the SIP user agent to block reception of SIP signaling messages on the TCP socket:

```
sip-ua
 no transport tcp
```

Related Commands

Command	Description
sip-ua	Enables the SIP user-agent configuration commands, with which you configure the user agent.

type (voice)

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

type {**1** | **2** | **3** | **5**}

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

Syntax Description

1	Indicates the following lead configuration: • E—Output, relay to ground. • M—Input, referenced to ground.
2	Indicates the following lead configuration: • E—Output, relay to SG. • M—Input, referenced to ground. • SB—Feed for M, connected to –48V. • SG—Return for E, galvanically isolated from ground.
3	Indicates the following lead configuration: • E—Output, relay to ground. • M—Input, referenced to ground. • SB—Connected to –48V. • SG—Connected to ground.
5	Indicates the following lead configuration: • E—Output, relay to ground. • M—Input, referenced to –48V.

Defaults

Type 1

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.
11.3(1)MA	This command was first supported on Cisco MC3810 multiservice concentrators.

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 type grounding the E-lead. The tie-line equipment detects the M-signal by detecting current flow to ground. If you select **1**, a common ground must exist between the line equipment and the PBX.

With **2**, the interface requires no common ground between the equipment, thereby avoiding ground loop noise problems. The E-signal is generated toward the PBX by connecting it to SG. The M-signal is indicated by the PBX connecting it to SB. While Type 2 interfaces do not require a common ground, they do have the tendency to inject noise into the audio paths because they are asymmetrical with respect to the current flow between devices.

Note E&M Type 4 is not a supported option. However, Type 4 operates similarly to Type 2 except for the M-lead operation. On Type 4, the M-lead states are open/ground, compared to Type 2, which is open/battery. Type 4 can interface with Type 2. To use Type 4 you can set the E&M voice port to Type 2 and perform the necessary M-lead rewiring.

With **3**, the interface operates the same as Type 1 interfaces with respect to the E-signal. The M-signal, however, is indicated by the PBX connecting it to SB on assertion and alternately connecting it to SG during inactivity. If you select **3**, a common ground must be shared between equipment.

With **5**, the Type 5 line equipment indicates E-signal to the PBX by grounding the E-lead. The PBX indicates M-signal by grounding the M-lead. A Type 5 interface is quasi-symmetrical in that while the line is up, current flow is more or less equal between the PBX and the line equipment, but noise injection is a problem.

Examples

The following example selects Type 3 as the interface type for the voice port on the Cisco 3600 series:

```
voice-port 1/0/0
 type 3
```

The following example selects Type 3 as the interface type for the voice port on the Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
 type 3
```

type (settlement)

To point to the provider type and the specific settlement server, use the **type** command in settlement configuration mode. To disable this command, use the **no** form of this command.

type {**osp** | **uni-osp**}

no type

Syntax Description

osp	Enables the Open Settlement Protocol (OSP) server type.
uni-osp	Enables authentication of Voice over IP (VoIP) calls to the Public Switched Telephone Network (PSTN) using a single settlement server.

Defaults

osp

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced on Cisco 2600 and 3600 series routers and Cisco AS5300 universal access servers.
12.1(2)T	The **uni-osp** keyword was introduced.

Usage Guidelines

This command defines the settlement server that is doing the accounting and enables the server to do the accounting.

Examples

The following example enables authentication of VoIP calls to the PSTN using a single settlement server:

```
settlement 0
 type uni-osp
```

Related Commands

Command	Description
connection-timeout	Sets the connection timeout.
customer-id	Sets the customer identification.
device-id	Sets the device identification.
encryption	Specifies the encryption method.
max-connection	Sets the maximum simultaneous connections.
response-timeout	Sets the response timeout.
retry-delay	Sets the retry delay.
retry-limit	Sets the connection retry limit.

Command	Description
session-timeout	Sets the session timeout.
settlement	Enters settlement configuration mode.
show settlement	Displays the configuration for all settlement server transactions.
shutdown/no shutdown	Brings up the settlement provider and then shuts it down.
url	Specifies the Internet service provider (ISP) address.

unbundle vfc

To unbundle DSPWare from the VCWare and configure the default file and capability lists with default values, use the **unbundle vfc** command in privileged EXEC mode.

unbundle [**high-complexity** | **medium-complexity**] **vfc** *slot-number*

Syntax Description

high-complexity	(Optional) Unbundles the high-complexity firmware set.
medium-complexity	(Optional) Unbundles the medium-complexity firmware set.
slot-number	Indicates the voice feature card (VFC) slot number.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(2)NA	This command was introduced on Cisco AS5300 universal access servers.
12.0(2)XH	The **high-complexity** and **medium-complexity** keywords were added.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.

Usage Guidelines

VFCs come with a single bundled image, VCWare, stored in VFC Flash memory. Use the **unbundle vfc** command to unbundle this bundled image into separate files, which are then written to Flash memory. When VCWare is unbundled, it automatically adds DSPWare to Flash memory, creates both the capability and default file lists, and populates these lists with the default files for that version of VCWare. The default file list includes the files that will be used to boot up the system. The capability list defines the available voice codecs for H.323 capability negotiation. These files are used during initial card configuration and for subsequent firmware upgrades.

Before unbundling a VFC software image that you have just copied over to VFC Flash, use the **clear vfc** command. Unbundling a DSP firmware set rewrites the default-file and capabilities lists. After unbundling, you must reload the router for any changes to take effect.

Examples

The following example unbundles the high-complexity firmware set into slot 2:

```
Router# unbundle high-complexity vfc 2
```

Related Commands

Command	Description
copy flash vfc	Copies a new version of VCWare from the Cisco AS5300 motherboard to VFC Flash memory.
copy tftp vfc	Copies a new version of VCWare from a TFTP server to VFC Flash memory.

url

To configure the Internet service provider (ISP) address, use the **url** command in settlement configuration mode. To disable this command, use the **no** form of this command.

url *url-address*

no url *url-address*

Syntax Description

url-address	Specifies the URL address. A valid URL address is as follows: http://*fully qualified domain name*[:port]**/**[URL]

Defaults

No default behavior or values.

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced on Cisco 2600 and 3600 series routers and Cisco AS5300 universal access servers.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

You can configure the address type multiple times. If you configure multiple URLs for the settlement server, the gateway attempts to send the request to each URL in the order in which you configured these addresses.

Examples

The following example configures four URLs for the settlement server:

```
settlement 0
 url http://1.2.3.4/
 url http://1.2.3.4:80/
 url https://1.2.3.4:4444/
 url https://yourcompany.com:443/
```

Related Commands

Command	Description
connection-timeout	Sets the connection timeout.
customer-id	Sets the customer identification.
device-id	Sets the device identification.
encryption	Specifies the encryption method.
max-connection	Sets the maximum simultaneous connections.
response-timeout	Sets the response timeout.
retry-delay	Sets the retry delay.

Command	Description
retry-limit	Sets the connection retry limit.
session-timeout	Sets the session timeout.
settlement	Enters settlement configuration mode.
show settlement	Displays the configuration for all settlement server transactions.
shutdown/no shutdown	Brings up the settlement provider and then shuts it down.
type	Specifies the provider type.

use-proxy

To enable proxy communications for calls between local and remote zones, use the **use-proxy** command in gatekeeper configuration mode. To either remove a proxy configuration entry for a remote zone or disable proxy communications between local and remote zones, use the **no** form of this command.

use-proxy *local-zone-name* {**default** | **remote-zone** *remote-zone-name*}{**inbound-to** | **outbound-from**}{**gateway** | **terminal**}

no use-proxy *local-zone-name* **remote-zone** *remote-zone-name* [{**inbound-to** | **outbound-from**}{**gateway** | **terminal**}]

Syntax Description

local-zone-name	The name or zone name of the gatekeeper, which is usually the fully domain-qualified host name of the gatekeeper. For example, if the domain name is cisco.com, the gatekeeper name might be gk1.cisco.com. However, if the gatekeeper is controlling multiple zones, the name of the gatekeeper for each zone should be a unique string that has a mnemonic value.
default	Defines the default proxy policy for all calls that are not defined by a **use-proxy** command with the **remote-zone** keyword.
remote-zone *remote-zone-name*	Defines a proxy policy for calls to or from a specific remote gatekeeper or zone.
inbound-to	Applies the proxy policy to calls that are inbound to the local zone from a remote zone. Each **use-proxy** command defines the policy for only one direction.
outbound-from	Applies the proxy policy to calls that are outbound from the local zone to a remote zone. Each **use-proxy** command defines the policy for only one direction.
gateway	Defines the type of local device to which the policy applies. The **gateway** option applies the policy only to local gateways.
terminal	Defines the type of local device to which the policy applies. The **terminal** option applies the policy only to local terminals.

Defaults

The local zone uses proxy for both inbound and outbound calls to and from the local H.323 terminals only. Proxy is not used for both inbound and outbound calls to and from local gateways.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
12.0(5)T	This command was introduced on Cisco AS5300 universal access servers.

Usage Guidelines

This command replaces the **zone access** command used in the previous versions of the gatekeeper. When a previous version of gatekeeper is upgraded, any **zone access** commands are translated to **use-proxy** commands. You can use the **show gatekeeper zone status** command to see the gatekeeper proxy configuration.

Examples

In the following example, the local zone sj.xyz.com is configured to use a proxy for inbound calls from remote zones tokyo.xyz.com and milan.xyz.com to gateways in its local zone. The sj.xyz.com zone is also configured to use a proxy for outbound calls from gateways in its local zone to remote zones tokyo.xyz.com and milan.xyz.com.

```
use-proxy sj.xyz.com remote-zone tokyo.xyz.com inbound-to gateway
use-proxy sj.xyz.com remote-zone tokyo.xyz.com outbound-from gateway
use-proxy sj.xyz.com remote-zone milan.xyz.com inbound-to gateway
use-proxy sj.xyz.com remote-zone milan.xyz.com outbound-from gateway
```

Because the default mode disables proxy communications for all gateway calls, only the gateway call scenarios listed above can use the proxy.

In the following example, the local zone sj.xyz.com uses a proxy for only those calls that are outbound from H.323 terminals in its local zone to the specified remote zone germany.xyz.com:

```
no use-proxy sj.xyz.com default outbound-from terminal
use-proxy sj.xyz.com remote-zone germany.xyz.com outbound-from terminal
```

Note Any calls inbound to H.323 terminals in the local zone sj.xyz.com from the remote zone germany.xyz.com use the proxy because the default applies.

The following example shows how to remove one or more proxy statements for the remote zone germany.xyz.com from the proxy configuration list:

```
no use-proxy sj.xyz.com remote-zone germany.xyz.com
```

The command above removes all special proxy configurations for the remote zone germany.xyz.com. After you enter a command like this, all calls between the local zone (sj.xyz.com) and germany.xyz.com are processed according to the defaults defined by any **use-proxy** commands that use the **default** option.

To prohibit proxy use for inbound calls to H.323 terminals in a local zone from a specified remote zone, enter a command similar to the following :

```
no use-proxy sj.xyz.com remote-zone germany.xyz.com inbound-to terminal
```

This command overrides the default and disables proxy use for inbound calls from remote zone germany.xyz.com to all H.323 terminals in the local zone sj.xyz.com.

Related Commands

Command	Description
show gatekeeper zone status	Displays the status of zones related to a gatekeeper.

vad (dial peer)

To enable voice activity detection (VAD) for the calls using a particular dial peer, use the **vad** command in dial-peer configuration mode. To disable VAD, use the **no** form of this command.

vad [aggressive]

no vad [aggressive]

Syntax Description

aggressive	Reduces noise threshold from –78 to –62 dBm. Available only when session protocol multicast is configured.

Defaults

VAD is enabled

Aggressive VAD is enabled in multicast dial peers

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series.
12.0(4)T	This command was implemented as a dial-peer command on Cisco MC3810 (in prior releases, the **vad** command was available only as a voice-port command).
12.2(10)	The **aggressive** keyword was added.

Usage Guidelines

Use this command to enable voice activity detection. With VAD, voice data packets fall into three categories: speech, silence, and unknown. Speech and unknown packets are sent over the network; silence packets are discarded. The sound quality is slightly degraded with VAD, but the connection monopolizes much less bandwidth. If you use the **no** form of this command, VAD is disabled and voice data is continuously sent to the IP backbone. When configuring voice gateways to handle fax calls, VAD should be disabled at both ends of the IP network because it can interfere with the successful reception of fax traffic.

When the **aggressive** keyword is used, the VAD noise threshold is reduced from –78 to –62 dBm. Noise that falls below the –62 dBm threshold is considered to be silence and is not sent over the network. Additionally, unknown packets are considered to be silence and are discarded.

On the Cisco MC3810, VAD can also be assigned to the voice port using the **vad (voice-port)** command. On the Cisco MC3810 multiservice concentrator, if you enable VAD on the dial peer for Voice over Frame Relay switched calls or permanent calls, the dial-peer setting overrides the VAD setting on the voice port.

Note On the Cisco MC3810, the **vad (dial-peer)** command is enabled by default. The **vad (voice-port)** command is disabled by default.

Examples

The following example enables VAD for a Voice over IP (VoIP) dial peer, starting from global configuration mode:

```
dial-peer voice 200 voip
 vad
```

Related Commands

Command	Description
comfort-noise	Generates background noise to fill silent gaps during calls if VAD is activated.
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
vad (voice-port)	Enables VAD for the calls using a particular voice port.

vad (voice-port)

To enable voice activity detection (VAD) for the calls using a particular voice port, use the **vad** command in voice-port configuration mode. To disable VAD, use the **no** form of this command.

vad

no vad

Syntax Description

This command has no arguments or keywords.

Defaults

VAD is not enabled.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced as a voice-port command on Cisco MC3810 multiservice concentrators.

Usage Guidelines

This command applies to Voice over Frame Relay and Voice over ATM on Cisco MC3810 multiservice concentrators.

Use the **vad** command to enable voice activity detection. With VAD, silence is not sent over the network; only audible speech is sent. If you enable VAD, the sound quality will be slightly degraded but the connection will monopolize much less bandwidth. If you use the **no** form of this command, VAD is disabled on the voice port.

Note It is recommended that you use the **vad** command in dial-peer configuration mode.

Examples

The following example enables VAD:

```
voice-port 1/1
 vad
```

Related Commands

Command	Description
comfort-noise	Generates background noise to fill silent gaps during calls if VAD is activated.
vad (dial peer)	Enables VAD for the calls using a particular dial peer.

vbr-rt

To configure the real-time variable bit rate (VBR) for Voice over ATM connections, use the **vbr-rt** command in ATM virtual circuit configuration mode. To restore the default, use the **no** form of this command.

vbr-rt *peak-rate average-rate burst*

no vbr-rt

Syntax Description

peak-rate	The peak information rate (PIR) of the voice connection, in kbps. The range is 56 to 10,000.
average-rate	The average information rate (AIR) of the voice connection, in kbps. The range is 1 to 56.
burst	Burst size, in number of cells. The range is 0 to 65,536.

Defaults

No vbr-rt settings are configured.

Command Modes

ATM virtual circuit configuration

Command History

Release	Modification
12.0	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(7)XK	Support for this command was extended to the Cisco 3600 series.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **vbr-rt** command configures traffic shaping between voice and data permanent virtual circuits (PVCs). Traffic shaping is required so that the carrier does not discard calls. To configure voice and data traffic shaping, you must configure the peak, average, and burst options for voice traffic. Configure the burst value if the PVC will be carrying bursty traffic. The peak, average, and burst values are needed so that the PVC can effectively handle the bandwidth for the number of voice calls. To calculate the *minimum* peak, average, and burst values for the number of voice calls, use the following calculations:

- Peak value: (2 x the maximum number of calls) x 16 kbps
- Average value: (1 x the maximum number of calls) x 16 kbps
- Burst value: (4 x the maximum number of calls)

Note When you configure data PVCs that will be traffic shaped with voice PVCs, use the **encapsulation aal5 snap** command and calculate the overhead as 1.13 times the voice rate.

Examples

The following example configures the traffic shaping rate for ATM PVC 20 on a Cisco 3600. In the example, the peak, average, and burst rates are calculated based on a maximum of 20 calls on the PVC.

```
pvc 20
 encapsulation aal5mux voice
 vbr-rt 640 56 80
```

Related Commands

Command	Description
encapsulation	Configures the ATM adaptation layer (AAL) and encapsulation type for an ATM PVC class.

vofr

To enable Voice over Frame Relay (VoFR) on a specific data-link connection identifier (DLCI) and to configure specific subchannels on that DLCI, use the **vofr** command in frame relay DLCI configuration mode. To disable VoFR on a specific DLCI, use the **no** form of this command.

Switched Calls

vofr [**data** *cid*] [**call-control** [*cid*]]

no vofr [**data** *cid*] [**call-control** [*cid*]]

Switched Calls to Cisco MC3810 Multiservice Concentrators Running Cisco IOS Releases Before 12.0(7)XK and 12.1(2)T

vofr [**cisco**]

no vofr [**cisco**]

Cisco-Trunk Permanent Calls

vofr data *cid* **call-control** *cid*

no vofr data *cid* **call-control** *cid*

Cisco-Trunk Permanent Calls to Cisco MC3810 Multiservice Concentrators Running Cisco IOS Releases Before 12.0(7)XK and 12.1(2)T

vofr cisco

no vofr cisco

FRF.11 Trunk Calls

vofr [**data** *cid*] [**call-control** *cid*]

no vofr [**data** *cid*] [**call-control** *cid*]

Syntax Description

data	(Required for Cisco-trunk permanent calls. Optional for switched calls.) Used to select a subchannel (CID) for data other than the default subchannel, which is 4.
cid	(Optional) Specifies the subchannel to be used for data. Valid values are from 4 through 255; the default is 4. If **data** is specified, enter a valid CID.
call-control	(Optional) Used to specify that a subchannel will be reserved for call-control signaling. This option is not supported on the Cisco MC3810 multiservice concentrator.

cisco	(Optional) Cisco proprietary voice encapsulation for VoFR with data is carried on CID 4 and call-control on CID 5. This option is required when configuring switched calls or Cisco trunks to Cisco MC3810 multiservice concentrators running Cisco IOS Releases before 12.0(7)XK and 12.1(2)T. If you are configuring switched calls or Cisco trunks to Cisco MC3810 multiservice concentrators running Cisco IOS Release 12.0(7)XK and 12.1(2)T and later releases, do not use this option.
cid	(Optional) Specifies the subchannel to be used for call-control signaling. Valid values are from 4 to 255; the default is 5. If **call-control** is specified and a CID is not entered, the default CID will be used.

Defaults

Disabled

Command Modes

Frame relay DLCI configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on Cisco 2600, 3600, and 7200 series routers and Cisco MC3810 multiservice concentrators.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.0(7)XK	The use of the **cisco** option was modified. Beginning in this release, use the **cisco** option only when configuring connections to Cisco MC3810 concentrators running Cisco IOS Releases before 12.0(7)XK and 12.1(2)T.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

Table 76 lists the different options of the **vofr** command and which combination of options is used beginning in Cisco IOS Release 12.0(7)XK and 12.1(2)T.

Table 76 Combinations of the vofr Command

Type of Call	Command Combination to Use
Switched call (user dialed or auto-ringdown) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** [*cid*]][1]
Switched call (user dialed or auto-ringdown) to a Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK and 12.1(2)T	**vofr cisco**[2]

Table 76 Combinations of the vofr Command (continued)

Type of Call	Command Combination to Use
Cisco-trunk permanent call (private-line) to other routers supporting VoFR	**vofr data** *cid* **call-control** *cid*
Cisco-trunk permanent call (private-line) to a Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK and 12.1(2)T	**vofr cisco**
FRF.11 trunk call (private-line) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** *cid*][3]

1. The recommended form of this command to use is **vofr data 4 call-control 5**.
2. This command consumes data CID 4 and call-control CID 5.
3. For FRF.11 trunk calls, the call-control option is not required. It is required only if you mix FRF.11 trunk calls with other types of voice calls on the same PVC.

Usage Restrictions for Cisco IOS Releases Before 12.0(7)XK and 12.1(2)T

This section describes restrictions for using the **vofr** command in releases before Cisco IOS Release 12.0(7)XK and 12.1(2)T. Beginning in Cisco IOS Release 12.0(7)XK and 12.1(2)T, these restrictions no longer apply.

When you use the **vofr** command without the **cisco** option, all subchannels on the DLCI are configured for FRF.11 encapsulation. If you enter the **vofr** command without any keywords or arguments, the data subchannel is CID 4 and there is no call-control subchannel.

Table 77 describes special conditions and restrictions for the use of the **vofr** command on the Cisco MC3810 running releases before 12.0(7)XK and 12.1(2)T.

Table 77 Using the vofr Command with the Cisco MC3810 Multiservice Concentrator

Type of Call	Conditions and Restrictions
FRF.11 trunks	1. Do *not* use the **cisco** option or the **call-control** option. 2. Use **vofr** or **vofr data** *cid*.
Cisco trunks	1. Must use **vofr cisco**.
switched-vofr	1. Must use **vofr cisco**.

If you select the “data” option, enter a numeric value to complete the command. If you select the **call-control** option, you do not enter a numeric value if you wish to accept the default call-control subchannel. See the following examples for clarification.

When you use the **vofr** command on a Cisco MC3810 multiservice concentrator without the “cisco” option, switched calls are not permitted. You can make only permanent FRF.11-trunk calls.

Note It is not possible to configure the **call-control** option on a Cisco MC3810 multiservice concentrator. If you configure this option, the setting is ignored.

Examples

The following example, beginning in global configuration mode, shows how to enable VoFR on serial interface 1/1, DLCI 100 on a Cisco 2600 series, 3600 series, or 7200 series router or on a Cisco MC3810 multiservice concentrator. The example configures CID 4 for data; no call-control CID is defined.

```
interface serial 1/1
 frame-relay interface-dlci 100
 vofr
```

To configure CID 4 for data and CID 5 for call-control (both defaults), enter the following command:

```
vofr call-control
```

To configure CID 10 for data and CID 15 for call-control, enter the following command:

```
vofr data 10 call-control 15
```

To configure CID 4 for data and CID 15 for call-control, enter the following command:

```
vofr call-control 15
```

To configure CID 10 for data and CID 5 for call-control, enter the following command:

```
vofr data 10 call-control
```

To configure CID 10 for data with no call-control, enter the following command:

```
vofr data 10
```

To configure a Cisco router or MC3810 for a VoFR application with an older release of the Cisco MC3810 (before Release 12.0(3)XG), enter the following command:

```
vofr cisco
```

Related Commands

Command	Description
class	Assigns a VC class to a PVC.
frame-relay interface-dlci	Assigns a DLCI to a specified Frame Relay subinterface.

voice call convert-discpi-to-prog

To convert a disconnect message with a progress indicator (PI) to a progress message, use the **voice call convert-discpi-to-prog** command in global configuration mode. To restore the default, use the **no** form of this command.

voice call convert-discpi-to-prog

no voice call convert-discpi-to-prog

Syntax Description

This command has no arguments or keywords.

Defaults

A disconnect message with a PI is not converted to a progress message.

Command Modes

Global configuration

Command History

Release	Modification
12.2(1)	This command was introduced.

Usage Guidelines

In Cisco IOS Release 12.2(1), a disconnect message with a PI is always converted to a progress message, even if the disconnect with PI is received after the connect message.

Examples

The following example changes a disconnect with PI to a progress message:

```
voice call convert-discpi-to-prog
```

Related Commands

Command	Description
disc_pi_off	Enables an H.323 gateway to disconnect a call when it receives a disconnect message with a PI.

voice call send-alert

To enable the terminating gateway to send an alert message instead of a progress message after it receives a call setup message, use the **voice call send-alert** command in global configuration mode. To restore the default behavior, use the **no** form of this command.

voice call send-alert

no voice call send-alert

Syntax Description

This command has no arguments or keywords.

Defaults

The terminating gateway sends a progress message after it receives a call Setup message.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)XI4	This command was introduced.
12.1(5)T	This command was not supported in this release.
12.1(5.3)T	This command was integrated into Cisco IOS Release 12.1(5.3)T.
12.2(1)	This command was integrated into Cisco IOS Release 12.2.

Usage Guidelines

In Cisco IOS Release 12.1(3)XI and later, the terminating gateway sends a Progress message with a progress indicator (PI) after it receives a Setup message. Previously, the gateway responded with an Alert message after receiving a call. In some cases, if the terminating switch does not forward the progress message to the originating gateway, the originating gateway does not cut-through the voice path until a Connect is received and the caller will not hear a ringback tone. In these cases, you can use the **voice call send-alert** command to make the gateway backward compatible with releases earlier than Cisco IOS Release 12.1(3)XI. If you configure the **voice call send-alert** command, the terminating gateway sends an Alert message after it receives a Setup message from the originating gateway.

To complete calls from a PRI to an FXS interface, configure the **voice call send-alert** command on the FXS device.

Examples

The following example configures the gateway to send an Alert message:

```
voice call send-alert
```

Related Commands

Command	Description
progress_ind	Sets a specific PI in call Setup, Progress, or Connect messages from an H.323 VoIP gateway.

voice-card

To configure a voice card, use the **voice-card** command in global configuration mode.

voice-card *slot*

Syntax Description

slot	On the Cisco 2600 and 3600 platforms, a value from 0 to 3 that identifies the physical slot in the chassis in which the voice card is located. On Cisco MC3810 mutliservice concentrators with one or two HCMs installed, enter 0 only; this applies to the entire chassis.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)XK	The command was introduced for the Cisco 2600 and 3600 series.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.
12.0(7)XK	This command was first supported on the Cisco MC3810 series.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

You can configure codec complexity only in voice-card configuration mode. On the Cisco 2600 and 3600 series, the slot corresponds to the physical slot in the chassis. On the Cisco MC3810 series, the slot is always 0, and all changes made in voice-card configuration mode apply to the entire Cisco MC3810. On Cisco MC3810 series concentrators, this command is available only if the chassis is equipped with one or two HCMs.

Examples

The following example enters voice-card configuration mode for the voice card in slot 1 on a Cisco 2600 or 3600 router:

```
voice-card 1
```

The following example enters voice-card configuration mode on a Cisco MC3810 concentrator:

```
voice-card 0
```

Related Commands

Command	Description
codec complexity	Matches the DSP complexity packaging to the codecs to be supported. Codec complexity changes are made in the voice-card configuration mode.

voice class busyout

To create a voice class for local voice busyout functions, use the **voice class busyout** command in global configuration mode. To delete the voice class, use the **no** form of this command.

voice class busyout *tag*

no voice class busyout *tag*

Syntax Description

tag	A unique identification number assigned to one voice class. The range is 1 to 10,000.

Defaults

No voice class is configured for busyout functions.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators.

Usage Guidelines

You can apply a busyout voice class to multiple voice ports. You can assign only one busyout voice class to a voice port. If a second busyout voice class is assigned to a voice port, the second voice class replaces the one previously assigned.

If you assign a busyout voice class to a voice port, you may not assign separate busyout commands directly to the voice port, such as **busyout monitor serial**, **busyout monitor ethernet**, or **busyout monitor probe**.

Examples

The following example configures busyout voice class 20, in which the connections to two remote interfaces are monitored by a response time reporter (RTR) probe with a G.711ulaw profile, and voice ports are busied out whenever both links have a packet loss exceeding 10 percent and a packet delay time exceeding 2 seconds:

```
voice class busyout 20
 busyout monitor probe 171.165.202.128 g711u loss 10 delay 2000
 busyout monitor probe 171.165.202.129 g711u loss 10 delay 2000
```

The following example configures busyout voice class 30, in which voice ports are busied out when serial ports 0/0, 1/0, 2/0, and 3/0 go out of service.

```
voice class busyout 30
 busyout monitor serial 0/0
 busyout monitor serial 1/0
 busyout monitor serial 2/0
 busyout monitor serial 3/0
```

Related Commands

Command	Description
busyout monitor ethernet	Configures a voice port to monitor a local Ethernet interface for events that would trigger a voice-port busyout.
busyout monitor probe	Configures a voice port to enter the busyout state if an RTR probe signal returned from a remote, IP-addressable interface crosses a specified delay or loss threshold.
busyout monitor serial	Configures a voice port to monitor a serial interface for events that would trigger a voice-port busyout.
show voice busyout	Displays information about the voice busyout state.

voice class codec

To enter voice-class configuration mode and assign an identification tag number for a codec voice class, use the **voice class codec** command in global configuration mode. To delete a codec voice class, use the **no** form of this command.

voice class codec *tag*

no voice class codec *tag*

Syntax Description

tag	The unique number you assign to the voice class. The valid range is 1 to 10,000. Each tag number must be unique on the router.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced on the Cisco AS5300.
12.0(7)T	This command was first supported on the Cisco 2600 and Cisco 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810 multiservice concentrator.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

This command only creates the voice class for codec selection preference and assigns an identification tag. Use the **codec preference** command to specify the parameters of the voice class, and use the **voice-class codec** dial-peer command to apply the voice class to a Voice over IP (VoIP) dial peer.

Note The **voice class codec** command in global configuration mode is entered without the hyphen. The **voice-class codec** command in dial-peer configuration mode is entered with the hyphen.

Examples

The following example shows how to enter voice-class configuration mode and assign a voice class tag number starting from global configuration mode:

```
voice class codec 10
```

After you enter voice-class configuration mode for codecs, use the **codec preference** command to specify the parameters of the voice class.

The following example creates preference list 99, which can be applied to any dial peer:

```
voice class codec 99
 codec preference 1 g711alaw
 codec preference 2 g711ulaw bytes 80
 codec preference 3 g723ar53
 codec preference 4 g723ar63 bytes 144
 codec preference 5 g723r53
 codec preference 6 g723r63 bytes 120
 codec preference 7 g726r16
 codec preference 8 g726r24
 codec preference 9 g726r32 bytes 80
 codec preference 10 g728
 codec preference 11 g729br8
 codec preference 12 g729r8 bytes 50
```

Related Commands

Command	Description
codec preference	Specifies a list of preferred codecs to use on a dial peer.
test voice port detector	Defines the order of preference in which network dial peers select codecs.
voice-class codec (dial peer)	Assigns a previously configured codec selection preference list to a dial peer.

voice-class codec (dial peer)

To assign a previously configured codec selection preference list (codec voice class) to a Voice over IP (VoIP) dial peer, enter the **voice-class codec** command in dial-peer configuration mode. To remove the codec preference assignment from the dial peer, use the **no** form of this command.

voice-class codec *tag*

no voice-class codec *tag*

Syntax Description

tag	The unique number assigned to the voice class. The valid range for this tag is 1 to 10,000. The *tag* number maps to the tag number created using the **voice class codec** global configuration command.

Defaults

Dial peers have no codec voice class assigned.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced on the Cisco AS5300.
12.0(7)T	This command was supported on the Cisco 2600 and 3600 series routers.
12.0(7)XK	This command was supported on the Cisco MC3810 multiservice concentrator.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

You can assign one voice class to each VoIP dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

Note The **voice-class codec** command in dial-peer configuration mode is entered with the hyphen. The **voice class codec** command in global configuration mode is entered without the hyphen.

Examples

The following example shows how to assign a previously configured codec voice class to a dial peer:

```
dial-peer voice 100 voip
 voice-class codec 10
```

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all dial peers configured on the router.
test voice port detector	Defines the order of preference in which network dial peers select codecs.
voice class codec	Enters voice-class configuration mode and assigns an identification tag number for a codec voice class.

voice class dualtone

To create a voice class for Foreign Exchange Office (FXO) supervisory disconnect tone detection parameters, use the **voice class dualtone** command in global configuration mode. To delete the voice class, use the **no** form of this command.

voice class dualtone *tag*

no voice class dualtone *tag*

Syntax Description

tag	A unique identification number assigned to one voice class. The range is from 1 to 10,000.

Defaults

No voice class is configured for tone detection parameters.

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators.

Usage Guidelines

Use this command first to create the voice class. Then use the **supervisory disconnect dualtone voice-class** command to assign the voice class to a voice port.

A voice class can define any number of tones to be detected. You need to define a matching tone for each supervisory disconnect tone expected from a PBX or from the public switched telephone network (PSTN).

Examples

The following example configures voice class dualtone 70, which defines one tone with two frequency components, and does not configure a cadence list:

```
voice class dualtone 100
 freq-pair 1 350 440
 freq-max-deviation 10
 freq-max-power 6
 freq-min-power 25
 freq-power-twist 15
 freq-max-delay 16
 cadence-min-on-time 50
 cadence-max-off-time 400
 cadence-variation 8
 exit
```

The following example configures voice class dualtone 100, which defines one tone with two frequency components, and configures a cadence list:

```
voice class dualtone 100
 freq-pair 1 350 440
 freq-pair 2 480 850
 freq-max-deviation 10
 freq-max-power 6
 freq-min-power 25
 freq-power-twist 15
 freq-max-delay 16
 cadence-min-on-time 50
 cadence-max-off-time 400
 cadence-list 1 100 100 300 300
 cadence-variation 8
 exit
```

The following example configures voice class dualtone 90, which defines three tones, each with two frequency components, and configures two cadence lists:

```
voice class dualtone 90
 freq-pair 1 350 440
 freq-pair 2 480 850
 freq-pair 3 1000 1250
 freq-max-deviation 10
 freq-max-power 6
 freq-min-power 25
 freq-power-twist 15
 freq-max-delay 16
 cadence-min-on-time 50
 cadence-max-off-time 500
 cadence-list 1 100 100 300 300 100 200
 cadence-list 2 100 200 100 400
 cadence-variation 8
 exit
```

Related Commands

Command	Description
supervisory disconnect dualtone voice-class	Assigns a previously configured voice class for FXO supervisory disconnect tone to a voice port.

voice class h323

To create an H.323 voice class that is independent of a dial peer and can be used on multiple dial peers, use the **voice class h323** command in global configuration mode. To remove the voice class, use the **no** form of this command.

voice class h323 *tag*

no voice class h323

Syntax Description

tag	Specifies a number to identify the voice class. The valid range for this tag is 1 to 10,000. The tag number must be unique on the router.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.1(2)T	This command was introduced on Cisco 1700 routers, Cisco 2600, 3600 and 7200 series routers, Cisco AS5300 universal access servers, and Cisco uBR910 and uBR924 cable access routers.

Usage Guidelines

The **voice class h323** command in global configuration mode does not include a hyphen. The **voice-class h323** command in dial-peer configuration mode uses a hyphen.

Examples

The following example creates an H.323 voice class labeled 1:

```
voice class h323 1
```

Related Commands

Command	Description
h225 timeout tcp establish	Sets the H.225 TCP timeout value.

voice-class h323 (dial peer)

To assign an H.323 voice class to a VoIP dial peer, use the **voice-class h323** command in dial-peer configuration mode. To remove the voice class from the dial peer, use the **no** form of this command.

voice-class h323 *tag*

no voice-class h323 *tag*

Syntax Description

tag	Specifies a number to identify the voice class. The valid range for this tag is 1 to 10,000. The tag number must be unique on the router.

Defaults

The dial peer does not use an H.323 voice class.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(2)T	This command was introduced.

Usage Guidelines

The voice class that you assign to the dial peer must be configured using the **voice class h323** in global configuration mode. The **voice-class h323** command in dial-peer configuration mode uses a hyphen and in global configuration mode does not include a hyphen.

You can assign one voice class to each VoIP dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

Examples

The following example shows how to create an H.323 voice class and then assign it to a dial peer:

```
voice class h323 10
dial-peer voice 100 voip
 voice-class h323 10
```

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all dial peers configured on the router.
voice class h323	Enters voice-class configuration mode and assigns an identification tag number for an H.323 voice class.

voice class permanent

To create a voice class for a Cisco trunk or FRF.11 trunk, use the **voice class permanent** command in global configuration mode. To delete the voice class, use the **no** form of this command.

voice class permanent *tag*

no voice class permanent *tag*

Syntax Description

tag	The unique number that you assign to the voice class. The *tag* number must be unique on the router. The valid range for this tag is 1 to 10,000.

Defaults

No voice class is configured.

Command Modes

Global configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 series.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

The **voice class permanent** command can be used for Voice over Frame Relay (VoFR), Voice over ATM (VoATM), and Voice over IP (VoIP) trunks.

Note The **voice class permanent** command in global configuration mode is entered without the hyphen. The **voice-class permanent** command in dial-peer and voice-port configuration modes is entered with the hyphen.

Examples

The following example shows how to create a permanent voice class starting from global configuration mode:

```
voice class permanent 10
 signal keepalive 3
 exit
```

Related Commands

Command	Description
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal timing oos	Configures the signal timing parameter for the OOS state of a call.
signal-type	Sets the signaling type for a network dial peer.
voice-class permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a network dial peer.

voice-class permanent (dial-peer)

To assign a previously configured voice class for a Cisco trunk or FRF.11 trunk to a network dial peer, use the **voice-class permanent** command in dial-peer configuration mode. To remove the voice-class assignment from the network dial peer, use the **no** form of this command.

voice-class permanent *tag*

no voice-class permanent *tag*

Syntax Description

tag	The unique number assigned to the voice class. The *tag* number maps to the tag number created using the **voice class permanent** global configuration command. The valid range is from 1 to 10,000.

Defaults

Network dial peers have no voice class assigned.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

You can assign one voice class to any given network dial peer. If you assign another voice class to a dial peer, the last voice class assigned replaces the previous voice class.

Note You cannot assign a voice class to a plain old telephone service (POTS) dial peer.

Note The **voice-class permanent** command in dial-peer configuration mode is entered with the hyphen. The **voice class permanent** command in global configuration mode is entered without the hyphen.

Examples

The following example assigns a previously configured voice class to a Voice over Frame Relay (VoFR) network dial peer:

```
dial-peer voice 100 vofr
 voice-class permanent 10
```

Related Commands

Command	Description
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal timing oos	Configures the signal timing parameter for the OOS state of a call.
signal-type	Sets the signaling type for a network dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.

voice-class permanent (voice-port)

To assign a previously configured voice class for a Cisco trunk or FRF.11 trunk to a voice port, use the **voice-class permanent** command in voice-port configuration mode. To remove the voice-class assignment from the voice port, use the **no** form of this command.

voice-class permanent *tag*

no voice-class permanent *tag*

Syntax Description

tag	The unique number assigned to the voice class. The *tag* number maps to the tag number created using the **voice class permanent** global configuration command. The valid range is from 1 to 10,000.

Defaults

Voice ports have no voice class assigned.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 multiservice concentrator.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was first supported as a voice port configuration command and on Cisco 2600 and 3600 series routers.

Usage Guidelines

You can assign one voice class to any given voice port. If you assign another voice class to a voice port, the last voice class assigned replaces the previous voice class.

Note The **voice-class permanent** command in voice-port configuration mode is entered with a hyphen. The **voice class permanent** command in global configuration mode is entered without the hyphen.

Examples

The following example assigns a previously configured voice class to voice port 1/1 in a Cisco MC3810 series concentrator:

```
voice-port 1/1
 voice-class permanent 10
```

The following example assigns a previously configured voice class to voice port 1/1/0 in a Cisco 3600 series router:

```
voice-port 1/1/0
 voice-class permanent 10
```

Related Commands

Command	Description
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Configures the signal timing parameter for the idle state of a call.
signal timing oos	Configures the signal timing parameter for the OOS state of a call.
signal-type	Sets the signaling type for a network dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.

voice confirmation-tone

To disable the two-beep confirmation tone for private line, automatic ringdown (PLAR) or PLAR off premises extension (OPX) connections, use the **voice confirmation-tone** command in voice-port configuration mode. To enable the two-beep confirmation tone, use the **no** form of this command.

voice confirmation-tone

no voice confirmation-tone

Syntax Description

This command has no arguments or keywords.

Defaults

The two-beep confirmation tone is heard on the PLAR or PLAR OPX connection.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on Cisco MC3810 multiservice concentrators.

Usage Guidelines

This command applies only to the Cisco MC3810 multiservice concentrator.

Use this command to disable the two-beep confirmation tone that a caller hears when picking up the handset for PLAR and PLAR OPX connections. This command is valid only if the voice-port **connection** command is set to PLAR or PLAR OPX.

Examples

The following example disables the two-beep confirmation tone on voice port 1/1 on the Cisco MC3810 multiservice concentrator:

```
voice-port 1/1
 connection plar-opx
 voice confirmation-tone
```

Related Commands

Command	Description
connection	Specifies a connection mode for a voice port.

voice-encap

This command was added in Cisco IOS Release 11.3(1)MA on the Cisco MC3810 multiservice concentrator. This command is not supported in Cisco IOS Release 12.2.

voice-group

This command was added in Cisco IOS Release 11.3(1)MA for the Cisco MC3810 multiservice concentrator. This command is not supported in Cisco IOS Release 12.2.

voice hunt

To configure an originating or tandem router so that it continues dial-peer hunting if it receives a user-busy disconnect code from a destination router, use the **voice hunt** command in global configuration mode. To configure the router so that it stops dial-peer hunting if it receives a user-busy disconnect code (the default option), use the **no** form of this command.

voice hunt {**user-busy** | **invalid-number** | **unassigned-number**}

no voice {**user-busy** | **invalid-number** | **unassigned-number**}

Syntax Description

user-busy	Router continues to dial-peer hunting if it receives a user-busy disconnect cause code from a destination router.
invalid-number	Router stops dial-peer hunting if it receives a an invalid-number disconnect cause code from a destination router.
unassigned-number	Router stops dial-peer hunting if it receives an unassigned-number disconnect cause code from a destination router.

Defaults

The default depends on the disconnect cause code. By default, the router stops dial-peer hunting if it receives the user-busy disconnect cause code. By default, the router continues dial-peer hunting if it receives an invalid-number, or an unassigned-number disconnect cause code.

Command Modes

Global configuration

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco 2600 series and 3600 series routers, and the Cisco MC3810 multiservice concentrator for Voice over Frame Relay (VoFR). It was also supported for Voice over IP (VoIP) on the 2600 series and 3600 series routers.
12.0(7)T	This command was first supported on the Cisco AS5300 and Cisco AS5800 for VoIP.
12.0(7)XK	This command was first used for VoIP on the Cisco MC3810 multiservice concentrator.
12.1(2)T	The support for VoIP on the Cisco MC3810 multiservice concentrator was implemented in Cisco IOS Release 12.1(2)T.
12.1(3)XI	The **invalid-number** and **unassigned-number** keywords were added, and the command name was changed to **voice hunt**.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.

Usage Guidelines

This command applies to routers that act as originating or tandem nodes in a Voice over IP, Voice over Frame Relay, or Voice over ATM environment.

This command is used for a configuration in which an originating or tandem router is configured with multiple dial peer entries that route a call to the same destination number, but on different destination routers. In this configuration, after all routes to the first router entry in the dial-peer list are active, a new call will not “roll over” to the next router in the dial-peer list.

This failure to route to the second destination router happens when the bandwidth on the voice interface is greater than the maximum capacity of the first destination router. This condition allows the originating or tandem router to attempt to place a new call to the first destination router because it has indications from the first destination router that there is more capacity based on the bandwidth setting. When the first destination router receives the call, if all of the ports are in use, the destination router returns a “user-busy” disconnect reason code to the originating or tandem router.

The originating or tandem router interprets the disconnect reason code as “unavailable destination” for the call and returns a busy tone to the initiating caller.

The originating or tandem router fails to try other routers in the dial-peer list after receiving a “user disconnect” reason code, and so it terminates the call attempt. By using this command, you can perform dial-peer hunting on multiple destination routers even if the originating or tandem router receives a “user-busy” disconnect reason code from one of the destination routers.

Examples

The following example displays configuring the originating or tandem router to continue dial-peer hunting if it receives a “user-busy” disconnect code from a destination router:

```
voice hunt user-busy
```

The following example displays configuring the originating or tandem router to continue dial-peer hunting if it receives an “invalid-number” disconnect code from a destination router:

```
voice hunt invalid-number
```

Related Commands

Command	Description
huntstop	Disables all further dial-peer hunting if a call fails when using hunt groups.
preference	Indicates the preferred order of a dial peer within a rotary hunt group.

voice local-bypass

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

voice local-bypass

no voice local-bypass

Syntax Description

This command has no arguments or keywords.

Defaults

Local calls bypass the DSP.

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced.
12.0(7)XK	This command was implemented on the Cisco 2600 and 3600 series routers and the MC3810 multiservice concentrator.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

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

Examples

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

```
no voice local-bypass
```

Related Commands

Command	Description
input gain	Configures a specific input gain value.
output attenuation	Configures a specific output attenuation value.

voice-port

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

Cisco 1750 Router

voice-port *slot-number/port*

Cisco 2600 and Cisco 3600 Series Router

voice-port {*slot-number***/***subunit-number***/***port*} | {*slot***/***port***:***ds0-group-no*}

Cisco MC3810 Multiservice Concentrator

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

Cisco AS5300 Universal Access Server

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

Cisco AS5800 Universal Access Server

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

Cisco 7200 Series Router

voice-port {*slot***/***port***:***ds0-group-no*} | {*slot-number***/***subunit-number***/***port*}

Syntax Description

For the Cisco 1750 Router:

slot-number	Slot number in the router in which the voice interface card (VIC) is installed. Valid entries are from 0 to 2, depending on the slot in which it has been installed.
port	Indicates the voice port. Valid entries are 0 or 1.

For the Cisco 2600 and Cisco 3600 Series Routers:

slot-number	Slot number in the Cisco router in which the VIC is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed.
subunit-number	Subunit on the VIC in which the voice port is located. Valid entries are 0 or 1.
port	Voice port number. Valid entries are 0 or 1.
slot	The router location in which the voice port adapter is installed. Valid entries are from 0 to 3.
port	Indicates the voice interface card location. Valid entries are 0 or 3.
dso-group-no	Indicates the defined DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

For the Cisco MC3810 Multiservice Concentrator:

slot/*port*	The *slot* argument specifies the slot number in the Cisco router in which the VIC is installed. The only valid entry is 1. The *port* variable specifies the voice port number. Valid ranges are as follows: • Analog voice ports: from 1 to 6. • Digital voice port: • Digital T1: from 1 to 24. • Digital E1: from 1 to 15, and from 17 to 31.

For the Cisco AS5300 Universal Access Server:

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

For the Cisco AS5800 Universal Access Server:

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

For the Cisco 7200 Series Router:

slot	The router location in which the voice port adapter is installed. Valid entries are from 0 to 3.
port	Indicates the VIC location. Valid entries are 0 or 1.
dso-group-no	Indicates the defined DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.
slot-number	Indicates the slot number in the Cisco router in which the VIC is installed. Valid entries are from 0 to 3, depending on the slot in which it has been installed.
subunit-number	Indicates the subunit on the VIC in which the voice port is located. Valid entries are 0 or 1.
port	Indicates the voice port number. Valid entries are 0 or 1.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(3)T	Support for Cisco 2600 series routers was added.
12.0(3)T	Support for the Cisco AS5300 access server was added.
12.0(7)T	Support for the Cisco AS5800 universal access server, the Cisco 7200 series router, and the Cisco 1750 router was added. Arguments for the Cisco 2600 and Cisco 3600 series router were added.

Usage Guidelines

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

Examples

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

```
voice-port 1/0/0
```

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

```
voice-port 1/24
```

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

```
voice-port 1:D
```

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

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

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

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

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice encapsulation.

voice-port busyout

To place all voice ports associated with a serial or ATM interface into a busyout state, use the **voice-port busyout** command in interface configuration mode. To remove the busyout state on the voice ports associated with this interface, use the **no** form of this command.

voice-port busyout

no voice-port busyout

Syntax Description

This command has no arguments or keywords.

Defaults

The voice ports on the interface are not in busyout state.

Command Modes

Interface configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on Cisco MC3810 multiservice concentrators.

Usage Guidelines

This command busies out all voice ports associated with the interface, except any voice ports configured to busy out under specific conditions using the **busyout monitor** and **busyout seize** commands.

Examples

The following example places the voice ports associated with serial interface 1 into busyout state:

```
interface serial 1
 voice-port busyout
```

The following example places the voice ports associated with ATM interface 0 into busyout state:

```
interface atm 0
 voice-port busyout
```

Related Commands

Command	Description
busyout forced	Forces a voice port on the Cisco MC3810 multiservice concentrator into the busyout state.
busyout monitor	Places a voice port on the Cisco MC3810 multiservice concentrator into the busyout monitor state.
busyout seize	Changes the busyout action for an FXO or FXS voice port.
show voice busyout	Displays information about the voice busyout state on the Cisco MC3810 multiservice concentrator.

voice rtp send-recv

To establish a two-way voice path when the Real-Time Transport Protocol (RTP) channel is opened, use the **voice rtp send-recv** command in global configuration mode. To restore the default condition, use the **no** form of this command.

voice rtp send-recv

no voice rtp send-recv

Syntax Description

This command has no arguments or keywords.

Defaults

The voice path is cut-through in only the backward direction when the RTP channel is opened.

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)T	This command was introduced on Cisco 2600, Cisco 3600, Cisco 7200, and Cisco 7500 series routers, Cisco AS5300 and Cisco AS5800 universal access servers, and Cisco MC3810 multiservice concentrators.

Usage Guidelines

The **voice rtp send-recv** command should be enabled only when the voice path must be cut-through (established) in both the backward and forward directions before a Connect message is received from the destination switch. The **voice rtp send-recv** command affects all Voice over IP (VoIP) calls when it is enabled.

Examples

The following example enables the voice path to cut-through in both directions when the RTP channel is opened:

```
voice rtp send-recv
```

voice service

To specify the voice encapsulation type, use the **voice service** command in global configuration mode.

voice service {**voip** | **voatm**}

Syntax Description

voip	Specifies Voice over IP (VoIP) parameters.
voatm	Specifies Voice over ATM (VoATM) parameters.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.1(1)XA	This command was introduced for VoATM for the Cisco MC3810 multiservice concentrators.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T for the Cisco MC3810 multiservice concentrators.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T for VoIP on the Cisco 2600 series routers, Cisco 3600 series routers, and Cisco MC3810 multiservice concentrators.

Usage Guidelines

Use the **voice service** command to switch to voice-service configuration mode from global configuration mode and to specify a voice encapsulation type. Use the **exit** command to exit the voice-service configuration mode and return to the global configuration mode.

Examples

The following example shows how to access voice-service configuration mode and specify VoIP parameters, beginning in global configuration mode:

```
voice service voip
```

Related Commands

Command	Description
fax protocol	Specifies the global default fax protocol for all the VoIP dial peers.
modem passthrough	Configures modem pass-through over VoIP.

voice vad-time

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

voice vad-time *milliseconds*

no voice vad-time

Syntax Description

milliseconds	The waiting period, in milliseconds, before silence detection and suppression of voice-packet transmission. The range is 250 to 65,536. The default is 250.

Defaults

250 milliseconds

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco 2600, 3600, and MC3810 multiservice concentrator.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **voice vad-time** command affects all voice ports on a router or concentrator, but it does not affect calls already in progress.

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

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

Examples

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

```
voice vad-time 20000
```

Related Commands

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

voip-incoming translation-rule

To set the incoming translation rule for calls that originate from H.323-compatible clients, use the **voip-incoming translation-rule** command in global configuration mode. To disable the incoming translation rule, use the **no** form of this command.

voip-incoming translation-rule *name-tag* {**calling-number** | **called-number**}

no voip-incoming translation-rule *name-tag* {**calling-number** | **called-number**}

Syntax Description

name-tag	The tag number by which the rule set will be referenced. This is an arbitrarily chosen number. The range is 1 through 2,147,483,647.
calling-number	The automatic number identification (ANI) number or the number of the calling party.
called-number	The Dial Number Information Service (DNIS) number or the number of the called party.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for Voice over IP on the Cisco AS5300.
12.0(7)XK	This command was first supported for Voice over IP on the Cisco 2600 series and Cisco 3600 series routers, and on the Cisco MC3810 multiservice concentrator.
12.1(1)T	This command was first supported on the T train for Voice over IP on the Cisco 1750, Cisco 2600 series, and Cisco 3600 series routers, the Cisco AS5300 universal access server, and the Cisco 7200 series and Cisco 7500 series routers.
12.1(2)T	This command was first supported on the T train for Voice over IP on the Cisco MC3810 multiservice concentrator.

Usage Guidelines

With this command, all IP-based calls are captured and handled, depending on either the calling number or the called number to the specified tag name.

Examples

The following example identifies the rule set for calls that originate from H.323-compatible clients:

```
voip-incoming translation-rule 5 called-number
```

Related Commands

Command	Description
numbering-type	Matches one number type for a dial-peer call leg.
rule	Applies a translation rule to a calling party number or a called party number for both incoming and outgoing calls.
show translation-rule	Displays the contents of all the rules that have been configured for a specific translation name.
test translation-rule	Tests the execution of the translation rules on a specific name-tag.
translate	Applies a translation rule to a calling party number or a called party number for incoming calls.
translate-outgoing	Applies a translation rule to a calling party number or a called party number for outgoing calls.
translation-rule	Creates a translation name and enters translation-rule configuration mode.

zone access

To configure the accessibility of your local-zone zone, use the **zone access** command in gatekeeper configuration mode. To remove any accessibility configurations, use the **no** form of this command.

zone access *local-zone-name* {**default** | **remote-zone** *remote-zone-name*}{**direct** | **proxied**}

no zone access *local-zone-name* **remote-zone** *remote-zone-name*

Syntax Description

local-zone-name	Name of local zone (synonymous with local gatekeeper).
default	Use with the **direct** or **proxied** keyword to define the mode of behavior for all remote zones that have not been specially named using the **remote-zone** *remote-zone-name* keyword and argument combination.
remote-zone *remote-zone-name*	Name of remote zone (synonymous with remote gatekeeper) for which a special mode of behavior is defined.
direct	Configures direct calls (without use of proxies) between endpoints. The local zone (or gatekeeper) offers the local endpoint IP address instead of the IP address of a local proxy.
proxied	Configures calls using proxies between endpoints. The local zone (or gatekeeper) offers the IP address of a local proxy instead of the local endpoint address.

Defaults

The local zone allows proxied access for all remote zones.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on Cisco 2500 and 3600 series routers.

Usage Guidelines

By default, a gatekeeper will offer a local proxy IP address when queried by a remote gatekeeper about a target local endpoint. This is considered proxied access. By using the **zone access** command, you can configure the local gatekeeper to offer the local endpoint address instead of the local proxy address. This is considered direct access.

Note The **zone access** command, configured on your local gatekeeper, affects onlythe use of proxies for incoming calls (that is, it does not affect the use of local proxies for outbound calls). When originating a call, a gatekeeper will use a proxy only if the remote gatekeeper offers a proxy at the remote end. A call between two endpoints in the same zone will always be a direct (nonproxied) call.

You can define the accessibility behavior of a local zone relative to certain remote zones using the **remote-zone** *remote-zone-name* keyword and argument combination with the **direct** or **proxied** keyword. You can define the default behavior of a local zone relative to all other remote zones using the **default** keyword with the **direct** or **proxied** keywords. To remove an explicitly named remote zone so that it is governed by the default-behavior rule, use the **no zone access** command.

Examples

The following example allows direct access to the local zone eng.xyz.com from remote zones within xyz corporation. All other remote locations will have proxied access to eng.xzy.com.

```
zone local eng.xyz.com xyz.com
zone access eng.xyz.com remote-zone mfg.xyz.com direct
zone access eng.xyz.com remote-zone mktg.xyz.com direct
zone access eng.xyz.com remote-zone sales.xyz.com direct
zone access eng.xyz.com default proxied
```

The following example supposes that only local gatekeepers within xyz.com have direct access to each other because your corporation has firewalls or you do not advertise your gatekeepers externally. You have excellent Quality of Service (QoS) within your corporate network, except for a couple of foreign offices. In this case, use proxies with the foreign offices (in Milan and Tokyo) and nowhere else.

```
zone local sanjose.xyz.com xyz.com
zone access sanjose.xyz.com default direct
zone access sanjose.xyz.com remote-zone milan.xyz.com proxied
zone access sanjose.xyz.com remote-zone tokyo.xyz.com proxied
```

Related Commands

Command	Description
show proxy h323 calls	Displays a list of each active call on the proxy.
zone local	Specifies a zone controlled by a gatekeeper.

zone bw

To set the maximum bandwidth allowed in a gatekeeper zone at any one time, use the **zone bw** command in gatekeeper configuration mode. To remove the maximum bandwidth setting and make the bandwidth unlimited, use the **no** form of this command.

zone bw *gatekeeper-name max-bandwidth*

no zone bw *gatekeeper-name max-bandwidth*

Syntax Description

gatekeeper-name	Name of the gatekeeper that controlsthe zone.
max-bandwidth	Maximum bidirectional bandwidth, in kbps, allowed in the zone at any one time.

Defaults

Bandwidth is unlimited.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on Cisco 2500 and 3600 series routers.

Examples

The following example sets the maximum bandwidth to 1000 kbps for zone gk1:

```
zone bw gk1 1000
```

Related Commands

Command	Description
show proxy h323 calls	Displays a list of each active call on the proxy.

zone local

To specify a zone controlled by a gatekeeper, use the **zone local** command in gatekeeper configuration mode. To remove a zone controlled by a gatekeeper, use the **no** form of this command.

zone local *gatekeeper-name domain-name* [*ras-IP-address*]

no zone local *gatekeeper-name domain-name*

Syntax Description

gatekeeper-name	The gatekeepers name or zone name. This is usually the fully domain-qualified host name of the gatekeeper. For example, if the *domain-name* is cisco.com, the *gatekeeper-name* might be gk1.cisco.com. However, if the gatekeeper is controlling multiple zones, the *gatekeeper-name* for each zone should be some unique string that has a mnemonic value.
domain-name	The domain name served by this gatekeeper.
ras-IP-address	(Optional) The IP address of one of the interfaces on the gatekeeper. When the gatekeeper responds to gatekeeper discovery messages, it signals the endpoint or gateway to use this address in future communications. **Note** Setting this address for one local zone makes it the address used for all local zones.

Defaults

No local zone is defined.

Note The gatekeeper cannot operate without at least one local zone definition. Without local zones, the gatekeeper goes to an inactive state when the **no shutdown** command is issued.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on Cisco 2500 and 3600 series routers.

Usage Guidelines

Multiple local zones can be defined. The gatekeeper manages all configured local zones. Intrazone and interzone behavior remains the same (zones are controlled by the same or different gatekeepers).

Only one *ras-IP-address* argument can be defined for all local zones. You cannot configure each zone to use a different RAS IP address. If you define this in the first zone definition, you can omit it for all subsequent zones, which automatically pick up this address. If you set it in a subsequent **zone local** command, it changes the RAS address of all previously configured local zones as well. Once defined, you can change it by reissuing any **zone local** command with a different *ras-IP-address* argument.

If the *ras-IP-address* argument is a Hot Standby Router Protocol (HSRP) virtual address, it automatically puts the gatekeeper into HSRP mode. In this mode, the gatekeeper assumes STANDBY or ACTIVE status according to whether the HSRP interface is on STANDBY or ACTIVE status.

You cannot remove a local zone if there are endpoints or gateways registered in it. To remove the local zone, shut down the gatekeeper first, which forces unregistration.

Multiple zones are controlled by multiple logical gatekeepers on the same Cisco IOS platform.

The maximum number of local zones defined in a gatekeeper should not exceed 100.

This command can also be used to change the IP address used by the gatekeeper.

Examples

The following example creates a zone controlled by a gatekeeper in the domain called cisco.com:

```
zone local gk1.cisco.com cisco.com
```

Related Commands

Command	Description
show proxy h323 calls	Displays a list of each active call on the proxy.
zone subnet	Specifies a zone controlled by a gatekeeper.

zone prefix

To add a prefix to the gatekeeper zone list, use the **zone prefix** command in gatekeeper configuration mode. To remove knowledge of a zone prefix, use the **no** form of this command with the gatekeeper name and prefix. To remove the priority assignment for a specific gateway, use the **no** form of this command with the **gw-priority** option.

zone prefix *gatekeeper-name e164-prefix* [**blast** | **seq**] [**gw-priority** *priority gw-alias* [*gw-alias, ...]*]

no zone prefix *gatekeeper-name e164-prefix* [**blast** | **seq**] [**gw-priority** *priority gw-alias* [*gw-alias, ...*]]

Syntax Description

gatekeeper-name	The name of a local or remote gatekeeper, which must have been defined by using the **zone local** or **zone remote** command.
e164-prefix	An E.164 prefix in standard form followed by dots (.). Each dot represents a number in the E.164 address. For example, 212....... is matched by 212 and any seven numbers. **Note** Although a dot representing each digit in an E.164 address is the preferred configuration method, you can also enter an asterisk (*) to match any number of digits.
blast	(Optional) If you list multiple hopoffs, this indicates that the LRQs should be sent simultaneously to the gatekeepers based on the order in which they were listed. The default is **seq**.
seq	(Optional) If you list multiple hopoffs, this indicates that the LRQs should be sent sequentially to the gatekeepers based on the order in which they were listed. The default is **seq**.
gw-priority *pri-0-to-10 gw-alias*	(Optional) Use the **gw-priority** option to define how the gatekeeper selects gateways in its local zone for calls to numbers beginning with prefix *e164-prefix*. Do not use this option to set priority levels for a prefix assigned to a remote gatekeeper. Use values from 0 to 10. A 0 value prevents the gatekeeper from using the gateway *gw-alias* for that prefix. Value 10 places the highest priority on gateway *gw-alias*. If you do not specify a priority value for a gateway, the value 5 is assigned. To assign the same priority value for one prefix to multiple gateways, list all the gateway names after the *pri-0-to-10* value. The *gw-alias* name is the H.323 ID of a gateway that is registered or will register with the gatekeeper. This name is set on the gateway with the **h323-gateway voip h.323-id** command.

Defaults

No knowledge of its own prefix or the prefix of any other zone is defined.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(6)Q	This command was introduced.
11.3(7)NA	This command was modified for H.323 Version 1.
12.0(5)T	The display format was modified for H.323 Version 2.

Usage Guidelines

A gatekeeper can handle more than one zone prefix, but a zone prefix cannot be shared by more than one gatekeeper. If you have defined a zone prefix as being handled by a gatekeeper and now define it as being handled by a second gatekeeper, the second assignment cancels the first.

If you need a gatekeeper to handle more than one prefix, but for cost reasons you want to be able to group its gateways by prefix usage, there are two ways to do it.

The first method is simpler, has less overhead, and is recommended if your gateways can be divided into distinct groups, in which each group is to be used for a different set of prefixes. For instance, if a group of gateways is used for calling area codes 408 and 650, and another group is used for calling area code 415, you can use this method. In this case, you define a local zone for each set of prefixes, and have the group of gateways to be used for that set of prefixes register with that specific local zone. Do not define any gateway priorities. All gateways in each local zone are treated equally in the selection process.

However, if your gateways cannot be cleanly divided into nonintersecting groups (for instance if one gateway is used for calls to 408 and 415 and another gateway is used for calls to 415 and 650, and so on), you can put all these gateways in the same local zone and use the **gw-priority** option to define which gateways will be used for which prefixes.

When choosing a gateway, the gatekeeper first looks for the longest zone prefix match; then it uses the priority and the gateway status to select from the gateways. If all gateways are available, the gatekeeper chooses the highest priority gateway. If all the highest priority gateways are busy (see the gateway **resource threshold** command), a lower priority gateway is selected.

Note The **zone prefix** command matches a prefix to a gateway. It does not register the gateway. The gateway must register with the gatekeeper before calls can be completed through that gateway.

Examples

The following example shows how you can define multiple local zones for separating your gateways:

```
zone local gk408or650 xyz.com
zone local gk415 xyz.com
zone prefix gk408or650 408.......
zone prefix gk408or650 650.......
zone prefix gk415 415.......
```

Now you need to configure all the gateways to be used for area codes 408 or 650 to register with gk408or650 and all gateways to be used for area code 415 to register with gk415. On Cisco voice gateways, you configure the gateways to register with the appropriate gatekeepers by using the **h323 voip id** command.

The following example shows how you can put all your gateways in the same zone but use the **gw-priority** keyword to determine which gateways will be used for calling different area codes:

```
zone local localgk xyz.com
```

```
zone prefix localgk 408.......
zone prefix localgk 415....... gw-priority 10 gw1 gw2
zone prefix localgk 650....... gw-priority 0 gw1
```

The commands shown accomplish the following tasks:

- Domain xyz.com is assigned to gatekeeper localgk.
- Prefix 408...... is assigned to gatekeeper localgk, and no gateway priorities are defined for it; therefore, all gateways registering to localgk can be used equally for calls to the 408 area code. No special gateway lists are built for the 408....... prefix; selection is made from the master list for the zone.
- The prefix 415....... is added to gatekeeper localgk, and priority 10 is assigned to gateways gw1 and gw2.
- Prefix 650....... is added to gatekeeper localgk, and priority 0 is assigned to gateway gw1.

A priority 0 is assigned to gateway gw1 to exclude it from the gateway pool for prefix 650........ When gateway gw2 registers with gatekeeper localgk, it is added to the gateway pool for each prefix as follows:

- For gateway pool for 415......., gateway gw2 is set to priority 10.
- For gateway pool for 650......., gateway gw2 is set to priority 5.

The following example changes gateway gw2 from priority 10 for zone 415....... to the default priority 5:

```
no zone prefix localgk 415....... gw-pri 10 gw2
```

The following example changes both gateways gw1 and gw2 from priority 10 for zone 415....... to the default priority 5:

```
no zone prefix localgk 415....... gw-pri 10 gw1 gw2
```

In the preceding example, the prefix 415....... remains assigned to gatekeeper localgk. All gateways that do not specify a priority level for this prefix are assigned a default priority of 5. The following example removes the prefix and all associated gateways and priorities from this gatekeeper:

```
no zone prefix localgk 415.......
```

Related Commands

Command	Description
register	Configures a gateway to register or deregister a fully qualified dial-peer E.164 address with a gatekeeper.
resource threshold	Configures a gateway to report H.323 resource availability to the gatekeeper of the gateway.
show call resource voice threshold	Displays the threshold configuration settings and status for an H.323 gateway.
show gateway	Displays the current gateway status.
zone local	Specifies a zone controlled by a gatekeeper.
zone remote	Statically specifies a remote zone if DNS is unavailable or undesirable.

zone remote

To statically specify a remote zone if domain name service (DNS) is unavailable or undesirable, use the **zone remote** command in gatekeeper configuration mode. To remove the remote zone, use the **no** form of this command.

zone remote *other-gatekeeper-name other-domain-name other-gatekeeper-ip-address* [*port-number*]

no zone remote *other-gatekeeper-name other-domain-name other-gatekeeper-ip-address* [*port-number*]

Syntax Description

other-gatekeeper-name	Name of the remote gatekeeper.
other-domain-name	Domain name of the remote gatekeeper.
other-gatekeeper-ip-address	IP address of the remote gatekeeper.
port-number	(Optional) RAS signaling port number for the remote zone. Value ranges from 1 to 65,535. If this is not set, the default is the well-known RAS port number 1719.

Defaults

No remote zone is defined. DNS will locate the remote zone.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on Cisco 2500 and 3600 series routers.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.

Usage Guidelines

All gatekeepers do not have to be in DNS. For those that are not, use the **zone remote** command so that the local gatekeeper knows how to access them. In addition, you may wish to improve call response time slightly for frequently accessed zones. If the **zone remote** command is configured for a particular zone, you do not need to make a DNS lookup transaction.

The maximum number of zones defined on a gatekeeper varies depending on the mode or the call model or both. For example, a directory gatekeeper may be in the mode of being responsible for forwarding location request (LRQ) messages and not handling any local registrations and calls; the call model might be E.164 addressed calls instead of H.323-ID addressed calls.

For a directory gatekeeper that does not handle local registrations and calls, the maximum remote zones defined should not exceed 10,000; an additional 4 MB of memory is required to store this maximum number of remote zones.

For a gatekeeper that handles local registrations and only E.164 addressed calls, the number of remote zones defined should not exceed 2000.

For a gatekeeper that handles H.323-ID calls, the number of remote zones defined should not exceed 200.

Examples

The following example configures the local gatekeeper to reach targets of the form *xxx*.cisco.com by sending queries to the gatekeeper named sj3.cisco.com at IP address 10.1.1.12:

```
zone remote sj3.cisco.com cisco.com 10.1.1.12
```

Related Commands

Command	Description
show proxy h323 calls	Displays a list of each active call on the proxy.
zone local	Specifies a zone controlled by a gatekeeper.

zone subnet

To configure a gatekeeper to accept discovery and registration messages sent by endpoints in designated subnets, use the **zone subnet** command in gatekeeper configuration mode. To disable the gatekeeper from acknowledging discovery and registration messages from subnets or to remove subnets entirely, use the **no** form of this command.

zone subnet *local-gatekeeper-name* {**default** | *subnet-address* {*/bits-in-mask* | *mask-address*}} **enable**

no zone subnet *local-gatekeeper-name* {**default** | *subnet-address* {*/bits-in-mask* | *mask-address*}} **enable**

Syntax Description

local-gatekeeper-name	Name of the local gatekeeper.
default	Applies to all other subnets that are not specifically defined by the **zone subnet** command.
subnet-address	Address of the subnet being defined.
/bits-in-mask	Number of bits of the mask to be applied to the subnet address.
mask-address	Mask (in dotted string format) to be applied to the subnet address.
enable	Gatekeeper accepts discovery and registration from the specified subnets.

Defaults

The local gatekeeper accepts discovery and registration requests from all subnets. If the request specifies a gatekeeper name, it must match the local gatekeeper name or the request will not be accepted.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on Cisco 2500 and 3600 series routers.
12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.

Usage Guidelines

You can use the **zone subnet** command more than once to create a list of subnets controlled by a gatekeeper. The subnet masks do not have to match actual subnets in use at your site. For example, to specify a particular endpoint, you can supply its address with a 32-bit netmask.

Examples

The following example starts by disabling the gatekeeper, gk1.cisco.com, from accepting discovery and registration messages from all subnets. Next, gk1.cisco.com is configured to accept discovery and registration messages from all H.323 nodes on the subnet 172.21.127.0.

In addition, gk1.cisco.com is configured to accept discovery and registration messages from a particular endpoint with the IP address 172.21.128.56.

```
no zone subnet gk1.cisco.com default enable
zone subnet gk1.cisco.com 172.21.127.0/24 enable
zone subnet gk1.cisco.com 172.21.128.56/32 enable
```

Related Commands

Command	Description
show gatekeeper zone status	Displays the status of zones related to a gatekeeper.
zone local	Specifies a zone controlled by a gatekeeper.