Cisco IOS Voice Commands: Si

This chapter contains commands to configure and maintain Cisco IOS voice applications. The commands are presented in alphabetical order. Some commands required for configuring voice 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 Configuration Guide*.

signal

To specify the type of signaling for a voice port, use the **signal** command in voice-port configuration mode. To reset to the default, 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**}

Centralized Automatic Message Accounting (CAMA) Ports

signal {**cama** {**kp–0–nxx–xxxx–st** | **kp–0–npa–nxx–xxxx–st** | **kp–2–st** | **kp–npd–nxx–xxxx–st**} | **groundstart** | **loopstart**}

no signal {**cama** {**kp–0–nxx–xxxx–st** | **kp–0–npa–nxx–xxxx–st** | **kp–2–st** | **kp–npd–nxx–xxxx–st**} | **groundstart** | **loopstart**}

Syntax Description

cama	Selects and configures the port for 911 calls.
delay-dial	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.
ground-start	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.
immediate	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.
kp–0–npa–nxx–xxxx–st	10-digit transmission. The E.164 number is fully transmitted.
kp–0–nxx–xxxx–st	7-digit ANI transmission. The Numbering Plan Area (NPA) or area code is implied by the trunk group and is not transmitted.
kp–2–st	Default transmission when the CAMA trunk cannot get a corresponding NPD digit in the look-up table, or when the calling number is fewer than 10 digits. (NPA digits are not available.)

kp-npd-nxx-xxx-st	8-digit ANI transmission, where the Numbering Plan Digit (NPD) is a single MF digit that is expanded into the NPA. The NPD table is preprogrammed in the sending and receiving equipment (on each end of the MF trunk); for example: 0= 415, 1=510, 2=650, 3=916 05551234 = (415) 555-1234, 15551234 = (510) 555-1234, and so on. NPD range is from 0 to 3.
loop-start	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.
wink-start	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.

Defaults

FXO and FXS interfaces: loop-start
E&M interfaces: wink-start
CAMA interfaces: loop-start

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.2(11)T	This command was modified to support ANI transmission.

Usage Guidelines

This command applies to analog voice ports only.

Using 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 take effect.

Some PBXs miss initial digits if the E&M voice port is configured for immediate signaling. If this occurs, use delay-dial signaling instead. Some 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.

To specify which VIC-2CAMA ports are designated as dedicated CAMA ports for emergency 911 calls, use the **signal cama** command. No two service areas in the existing North American telephony infrastructure supporting E911 calls have identical service implementations, and many of the factors that drive the design of emergency call handling are matters of local policy and therefore outside the scope of this document. Local policy determines which ANI format is appropriate for the specified PSAP location.

The following four types of ANI transmittal schemes are based on the actual number of digits transmitted toward the E911 tandem or the PSAP. In each instance, the actual calling number is proceeded with a key pulse (KP) followed by an information field (I) or a Numbering Plan Digit (NPD), which is then followed by the Automatic Number Identification (ANI) calling number, and finally followed by a start pulse (ST). The value 0 indicates that the single digit information field (I). ANI was provided by the private branch exchange (PBX). The value 2 indicates that ANI was not provided by the PBX.

- **7-digit transmission (kp-0-nxx-xxxx-st*)***:

  The calling phone number is transmitted, and the NPA is implied by the trunk group and not transmitted.

- **8-digit transmission** (**KP-npd-nxx-xxxx-st**):

  The I field consists of single-digit NPD-to-NPA mapping. When the calling party number of 415-555-2222 places a 911 call, and the Cisco 2600 series or Cisco 3600 series has an NPD (0) to NPA (415) mapping, the NPA signaling format is received by the selective router at the central office (CO).

  Note NPD values greater than 3 are reserved for signifying error conditions.

- **10-digit transmission (kp-0-npa-nxx-xxxx-st)**:

  The E.164 number is fully transmitted.

- **kp–2–st transmission** (**kp–2–st**):

  kp–2–st transmission is used if the PBX is unable to out-pulse the ANI. If the ANI received by the Cisco router is not as per configured values, kp–2–st is transmitted. For example, if the voice port is configured for outpulsing a 10-digit ANI and the 911 call it receives has a 7-digit calling party number, the router transmits kp–2–st.

  Note Emergency 911 calls are not rejected for an ANI mismatch. The call establishes a voice path. The E911 network, however, does not receive the ANI.

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
```

The following example configures a 10-digit ANI transmission:

```
Router(config)# voice-port 1/0/0
Router(config-voiceport)# signal cama kp-0-npa-nxxx-xxxx-st
```

Related Commands

Command	Description
ani mapping	Preprograms the NPA, or area code, into a single MF digit.
voice–port	Enters voice-port configuration mode.

signal did

To enable direct inward dialing (DID) on a voice port, use the **signal did** command in voice-port configuration mode. To disable DID and reset to loop-start signaling, use the **no** form of this command.

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

no signal did

Syntax Description

immediate-start	Enables immediate-start signaling on the DID voice port.
wink-start	Enables wink-start signaling on the DID voice port.
delay-start	Enables delay-dial signaling on the DID voice port.

Defaults

No default behavior or values

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(5)XM	This command was introduced on the Cisco 2600 series and Cisco 3600 series.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco IAD2420 series.

Examples

The following example configures a voice port on a Cisco IAD2420 series IAD with immediate-start signaling enabled:

```
Router# voice-port 1/17
Router (config-voiceport)# signal did immediate-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 reset to the default, use the **no** form of this command.

signal keepalive *number*

no signal keepalive *number*

Syntax Description

number	Keepalive signaling packet interval, in seconds. Range is from 1 to 65535.

Defaults

5 seconds

Command Modes

Voice-class configuration

Command History

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

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 patterns 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	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	Signaling pattern for identifying an idle message from the PBX.
oos receive	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	Signaling pattern for identifying an OOS message from the PBX.
bit-pattern	ABCD bit pattern. Range is from 0000 to 1111.

Defaults

idle receive	Near-end E&M: 0000 (for T1) or 0001 (for E1) Near-end FXO loop start: 0101 Near-end FXO ground start: 1111 Near-end FXS: 0101 Near-end MELCAS: 1101
idle transmit	Near-end E&M: 0000 Near-end FXO: 0101 Near-end FXS loop start: 0101 Near-end FXS ground start: 1111 Near-end MELCAS: 1101
oos receive	Near-end E&M: 1111 Near-end FXO loop start: 1111 Near-end FXO ground start: 0000 Near-end FXS loop start: 1111 Near-end FXS ground start: 0101 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.
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 implemented on the Cisco 2600 series and Cisco 3600 series.

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 are received, the router sends 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 nondefault 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 voice-class configuration mode. To reset to the default, 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.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.

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 signal 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 a call, use the s**ignal timing idle suppress-voice** command in voice-class configuration mode. To reset to the default, 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. Range is from 0 to 65535.

Defaults

No signal timing idle suppress-voice timer is configured.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810.
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 implemented on the Cisco 2600 series and Cisco 3600 series.

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 reset to the default, 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 is torn down and an attempt to achieve reconnection is 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. Range is from 0 to 65535.

Defaults

No signal timing OOS pattern parameters are configured.

Command Modes

Voice-class configuration

Command History

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

Usage Guidelines

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

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

Examples

The following example, beginning in global configuration mode, configures the signal timeout parameter for the 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 reset to the default, 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. Range is from 0 to 65535. There is no default.

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.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.

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 reset to the default, 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. Range is from 0 to 65535. There is no default.

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.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.

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
```

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-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 reset to the default, 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. Range is from 0 to 65535. There is no default.

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.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.

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 reset to the default, 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. Range is from 0 to 65535. There is no default.

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.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.

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 reset to the default, 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. Range is from 1 to 65535. Default is 30.
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.
12.0(4)T	This command was integrated into Cisco IOS Release 12.0(4)T.
12.1(3)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.

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.

signaling forward

To enable a Cisco IOS gateway to forward the Generic Transparency Descriptor (GTD) payload to another gateway or gatekeeper system-wide, use the **signaling forward** command in global configuration mode. To disable forwarding, use the **no** form of this command.

signaling forward {**conditional** | **unconditional** | **none**}

no signaling forward

Syntax Description

conditional	Changes the forwarding behavior on the basis of the target defined in the **session target** command. If the target is a non-Registration, Admission, and Status (RAS) target, the original signaling payload is forwarded to the H.323 endpoint using H.225 messages.
unconditional	Tunnels the GTD payload in the H.225 SETUP message to the final endpoint in the network. The gatekeeper sends its own GTD back to itself in this situation.
none	Prevents the gateway from forwarding the GTD payload to endpoints in the network.

Defaults

Signaling forwarding is conditional.

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco AS5350 and Cisco AS5850.

Usage Guidelines

This command is used with the Cisco PGW 2200 in the Cisco SS7 Interconnect for Voice Gateways solution. You must configure the Cisco PGW 2200 to encapsulate SS7 ISUP messages in GTD format before using this command on the Cisco gateway.

If the target is a RAS target, for a non-GTD signaling payload, the original payload is forwarded. For a GTD signaling payload, the payload is encapsulated in an admission request (ARQ)/disengage request (DRQ) message and sent to the originating gatekeeper. The gatekeeper conveys the payload to the Gatekeeper Transaction Message Protocol (GKTMP) and external route server for a flexible route decision based upon the ISDN User Part (ISUP) GTD parameters. The gateway then conditionally forwards the GTD payload on the basis of the instruction from the route server.

This command does not prevent sending the GTD to a gatekeeper. Any GTD on the originating gateway is sent to the gatekeeper for use in routing decisions. To prevent GTD creation, the **signal-end-to-end** command-line interface (CLI) option on the R2 interfaces should be disabled, and the Cisco PGW 2200 should be configured not to send GTD to the gateway.

Examples

The following example sets unconditional signal forwarding on a system-wide basis, where the GTD payload is tunneled in H.225 SETUP messages to endpoints:

```
Router(config)# voice service voip
Router(conf-voi-serv)# signaling forward unconditional
Router(conf-voi-serv)# ^Z
Router# show running-config

Building configuration...

Current configuration : 4201 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
hostname as5300-2
!
no logging buffered
logging rate-limit console 10 except errors
aaa new-model
!
.
.
.
!
voice service voip
 signaling forward unconditional
 h323
!
.
.
.
```

Related Commands

clid network-number	Configures a network number in the router for CLID and uses it as the calling party number.
clid restrict	Prevents the calling party number from being presented by CLID.
clid second-number strip	Prevents the second network number from being sent in the CLID information.
session target	Specifies a network-specific address for a dial peer.

signaling forward (dial-peer)

To enable a Cisco IOS gateway to forward the Generic Transparency Descriptor (GTD) payload to another gateway or gatekeeper for an individual dial peer, use the **signaling forward** command in dial-peer configuration mode. To disable forwarding, use the **no** form of this command.

signaling forward {**conditional** | **unconditional** | **none**}

no signaling forward

Syntax Description

conditional	Changes the forwarding behavior on the basis of the target defined in the **session target** command. If the target is a non-Registration, Admission, and Status (RAS) target, the original signaling payload is forwarded to the H.323 endpoint using H.225 messages.
unconditional	Tunnels the GTD payload in the H.225 SETUP message to the final endpoint in the network. The gatekeeper sends its own GTD back to itself in this situation.
none	Prevents the gateway from passing the GTD payload to endpoints in the network.

Defaults

The default is the value that is configured system-wide, or conditional if signaling forward is not configured system-wide.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco AS5350 and Cisco AS5850.

Usage Guidelines

This command is used with the Cisco PGW 2200 Signaling Controller in the Cisco SS7 Interconnect for Voice Gateways solution. You must configure the Cisco PGW 2200 to encapsulate SS7 ISUP messages in GTD format before using this command on the Cisco gateway.

If the target is a RAS target, for a non-GTD signaling payload, the original payload is forwarded. For a GTD signaling payload, the payload is encapsulated in an admission request (ARQ)/disengage request (DRQ) message and sent to the originating gatekeeper. The gatekeeper conveys the payload to the Gatekeeper Transaction Message Protocol (GKTMP) and external route server for a flexible route decision based upon the ISDN User Part (ISUP) GTD parameters. The gateway then conditionally forwards the GTD payload on the basis of the instruction from the route server.

This command does not prevent sending the GTD to a gatekeeper. Any GTD on the originating gateway is sent to the gatekeeper for use in routing decisions. To prevent GTD creation, the **signal-end-to-end** command-line interface (CLI) option on the R2 interfaces should be disabled, and the Cisco PGW 2200 should be configured not to send GTD to the gateway.

Examples

The following example sets unconditional signal forwarding on a system-wide basis, where the GTD payload is tunneled in H.225 SETUP messages to endpoints:

```
Router(config)# voice service voip
Router(conf-voi-serv)# signaling forward unconditional
Router(conf-voi-serv)# ^Z
Router# show running-config

Building configuration...

Current configuration : 4201 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
hostname as5300-2
!
no logging buffered
logging rate-limit console 10 except errors
aaa new-model
!
.
.
.
!
voice service voip
 signaling forward unconditional
 h323
!
.
.
.
```

Related Commands

clid network-number	Configures a network number in the router for CLID and uses it as the calling party number.
clid restrict	Prevents the calling party number from being presented by CLID.
clid second-number strip	Prevents the second network number from being sent in the CLID information.
session target	Specifies a network-specific address for 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 reset to the default, 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, 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 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 the Cisco 2600, Cisco 3600, and Cisco MC3810.
12.0(4)T	This command was implemented on the Cisco 7200 series.
12.0(7)XK	The **cept** and **transparent** keywords, previously supported only on the Cisco MC3810, are now supported on the Cisco 2600 series, Cisco 3600 series, and 7200 series.
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 and Cisco 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 disables signaling on a Cisco 2600series or Cisco 3600 series router or on a Cisco MC3810 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.

silent-fax

To configure the voice dial peer for a Type 2 silent fax machine, use the **silent-fax** command in dial-peer voice configuration mode. To disable a silent fax call to any POTS ports, use the **no** form of this command.

silent-fax

no silent-fax

Syntax Description

This command has no arguments or keywords.

Defaults

Silent fax is not configured.

Command Modes

Dial-peer voice configuration

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco 803, Cisco 804, and Cisco 813.

Usage Guidelines

Use this command to configure the router to send a no ring alert tone to a Type 2 silent fax machine that is connected to any of the POTS ports. To check the status of the silent-fax configuration, use the **show running-config** command.

Examples

The following example shows that the **silent-fax** command has been configured on POTS port 1 but not on POTS port 2.

```
dial-peer voice 1 pots
 destination-pattern 5551111
 port 1
 no call-waiting
 ring 0
 volume 4
 caller-number 3334444 ring 1
 subaddress 20
 silent-fax

dial-peer voice 2 pots
 destination-pattern 5552222
 port 2
 no call-waiting
 ring 0
 volume 2
 caller-number 3214567 ring 2
 subaddress 10
```

sip

To enter SIP configuration mode, use the **sip** command in voice-service VoIP configuration mode.

sip

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Voice-service VoIP configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400.
12.2(2)XB2	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.

Usage Guidelines

From the voice-service VoIP configuration mode, this command enables you to enter SIP configuration mode. From this mode, several SIP commands are available, such as the **bind**, **session transport**, and **url**. commands.

Examples

The following example enters SIP configuration mode and sets the **bind** command on the SIP network:

```
Router(config)# voice service voip
Router(config-voi-srv)# sip
Router(conf-serv-sip)# bind control source-interface FastEthernet 0
```

Related Commands

Command	Description
session transport	Configures the voice dial peer to use TCP or UDP as the underlying transport layer protocol for SIP messages.
voice service voip	Enters voice-service configuration mode.

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. To remove a network address configured for SIP, use the **no** form of this command.

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

no sip-server {**dns:**[*host-name*] | **ipv4:***ip-addr* [**:***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) Valid DNS host name in 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) Port number for the SIP server.

Defaults

Null value

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

Usage Guidelines

If you use this command, you can also use the **session target** {**sip-server}** command on each dial peer instead of repeatedly entering the SIP server interface address for each dial peer. Configuring a SIP server as a session target is useful if there is a Cisco SIP proxy server (CSPS) present in the network. With a CSPS, you can configure the SIP server option and have the interested dial peers use the CSPS by default.

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 “3660-2.sip.com.” If you also use the command **session target {sip server}**, you do not have to set the DNS host name for each individual dial peer.

```
sip-ua
 sip-server dns:3660-2.sip.com

dial-peer voice 29 voip
 session target sip-server
```

The following example sets the global SIP server interface to an IP address:

```
sip-ua
 sip-server ipv4:10.0.2.254
```

Related Commands

Command	Description
ip name-server	Specifies the address of one or more name servers to use for name and address resolution.
session target (VoIP)	Specifies a network-specific address for a dial peer.
sip-ua	Enters SIP user-agent configuration mode in order to configure the SIP user agent.

sip-ua

To enable Session Initiation Protocol (SIP) user-agent configuration commands, in order to configure the user agent, use the **sip-ua** command in global configuration mode. To reset all SIP user-agent configuration commands to their default values, use the **no** form of this command.

sip-ua

no 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 the Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.

Usage Guidelines

Use this command to enter SIP user-agent configuration mode. Table 153 lists the SIP user-agent configuration mode commands:

Table 153 SIP User-Agent Configuration Mode Commands

Command	Description
exit	Exits SIP user-agent configuration mode.
inband-alerting	This command is no longer supported as of Cisco IOS Release 12.2. This command is no longer needed because the gateway handles remote or local ringback on the basis of SIP messaging.
max-forwards	Specifies the maximum number of hops for a request.
retry	Configures the SIP signaling timers for retry attempts.
sip-server	Configures a SIP server interface.

Table 153 SIP User-Agent Configuration Mode Commands (continued)

Command	Description
timers	Configures the SIP signaling timers.
transport	Enables or disables a SIP user agent transport for TCP or UDP that the protocol SIP user agents listen 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, and 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.
max-forwards	Specifies the maximum number of hops for a request.
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 notifications for applicable calls associated with VoIP dial peers, use the **snmp enable peer-trap poor-qov** command in dial-peer configuration mode. To disable 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 the Cisco 3600 series.

Usage Guidelines

Use this command to generate poor-quality-of-voice notification for applicable calls associated with a 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 notification 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.

speed-dial (ephone)

To set speed-dial buttons on a Cisco IP phone, use the **speed-dial** command in ephone configuration mode. To disable the buttons, use the **no** form of this command.

speed-dial *button-number directory-number*

no speed-dial *button-number directory-number*

Syntax Description

button-number	Speed-dial string tag for the Cisco IP phone speed-dial button number. Range is from 1 to 4.
directory-number	Directory number on a phone.

Defaults

No default behavior or values

Command Modes

Ephone configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

If more speed-dial buttons are defined than are actually supported by the physical phone, the extra speed-dial configurations are ignored.

Examples

The following example sets speed-dial button 1 for directory number 5001:

```
Router(config)# ephone 1
Router(config-ephone)# speed-dial 1 5001
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.

srv version

To generate Domain Name System Server (DNS SRV) queries with either RFC 2052 or RFC 2782 format, use the **srv version** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

srv version {**1** | **2**}

no srv version

Syntax Description

1	Domain-name prefix of format protocol.transport. (RFC 2052 style).
2	Domain-name prefix of format _protocol._transport. (RFC 2782 style).

Defaults

2 (RFC 2782 style)

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5850 is not included in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. This command is supported on the Cisco AS5850 in this release.

Usage Guidelines

Session Initiation Protocol (SIP) on Cisco VoIP gateways uses DNS SRV queries to determine the IP address of the user endpoint. The query string has a prefix in the form of "protocol.transport." (RFC 2052) or "_protocol._transport." (RFC 2782). The selected string is then attached to the fully qualified domain name (FQDN) of the next hop SIP server.

By choosing the value of 1, this command provides compatibility with older equipment that supports only RFC 2052.

Examples

The following example sets up the **srv version** command in the RFC 2782 style (underscores surrounding the protocol):

```
Router(config)# sip-ua
Router(config-sip-ua)# srv version 2
```

Related Commands

Command	Description
show sip-ua status	Displays SIP status.

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 mtp2-variant bellcore [*channel*] [*parameters*]

Syntax Description

channel	(Optional) Channel. Range is from 0 to 3.
parameters	(Optional) Particular Bellcore standard. See Table 154 for descriptions, defaults, and ranges.

Defaults

Bellcore is the default variant if no other is configured.
See Table 154 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 154. 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 an actual timeout duration of 1000 ms. This is true for all timers and all variants.

Table 154 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
lssu-len	1- or 2-byte LSSU format	1	1 to 2

Table 154 Bellcore (Telcordia Technologies) 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
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 ms:

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

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

```
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	Channel. Range is from 0 to 3.
parameters	(Optional) Particular Bellcore standard. See Table 155 for descriptions, defaults, and ranges.

Defaults

Bellcore is the default variant if no other is configured.
See Table 155 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 155. To restore the designated default, use the **no** or the **default** form of the command (see the example below).

Table 155 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 155 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 ms:

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

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

```
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	Channel. Range is from 0 to 3.
parameters	(Optional) Particular Bellcore standard. See Table 156 for descriptions, defaults, and ranges.

Defaults

Bellcore is the default variant if no other is configured.
See Table 156 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 156. To restore the designated default, use the **no** or the **default** form of the command (see the example below).

Table 156 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 156 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	Channel. Range is from 0 to 3.
parameters	(Optional) Particular Bellcore standard. See Table 157 for descriptions, defaults, and ranges.

Defaults

Bellcore is the default variant if no other is configured.
See Table 157 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 157. To restore the designated default, use the **no** or the **default** form of the command (see the example below).

Table 157 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 157 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 and explicitly add an RUDP session to a Signaling System 7 (SS7) session set, use the **ss7 session** command in global configuration mode. To delete the session, use the **no** form of this command.

ss7 session *session-id* **address** *destination-address destinaion-port local-address local-port* [**session-set** *session-number*]

no ss7 session *session-id*

Syntax Description

session-id	SS7 session number. Valid values are 0 and 1. You must enter a hyphen with no space following it after the **session** keyword.
address *destination-address*	Specifies the SS7 session IP address.
destination-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.
destination-port	The number of the local UDP port on which the router expects to receive messages from the media gateway controller (MGC). 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. Valid port ranges are from 1024 to 9999.
local-address	The remote IP address of the MGC in four-part dotted-decimal format.
local-port	The number of the remote UDP port on which the MGC 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. Valid port ranges are from 1024 to 9999.
session-set *session-number*	(Optional) Assigns an SS7 session to an SS7 session set.

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.
12.2(15)T	The **session-set** keyword and the *session-number* argument were added.

Usage Guidelines

For the Cisco 2600-based SLT, you can configure a maximum of four sessions, two for each Cisco SLT. In a redundant VSC configuration, session 0 and session 2 are configured to one VSC, and session 1 and session 3 are configured to the other. Session 0/1 and session 2/3 run to the Cisco SLT.

The VSC must be configured to send messages to the local port, and it must be configured to listen on the remote port. You must also reload the router whenever you remove a session or change the parameters of a session.

This command replaces the **ss7 session-0 address** and **ss7 session-1 address** commands, which contain hard-coded session numbers. The new command is used for the new dual Ethernet capability.

The new CLI supports both single and dual Ethernet configuration by being backward compatible with the previous **session-0** and **session-1** commands so that you can configure a single Ethernet instead of two, if needed.

For the Cisco AS5350 and Cisco AS5400-based SLT, you can configure a maximum of two sessions, one for each signaling link. In a redundant MGC configuration, session 0 is configured to one MGC and session 1 is configured to the other.

The MGC must be configured to send messages to the local port, and the MGC 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.

By default, each RUDP session must belong to SS7 session set 0. This allows backward compatibility with existing SS7 configurations.

If the **session-set** keyword is omitted, the session is added to the default SS7 session set 0. This allows backward compatibility with older configurations. Entering the **no** form of the command is still sufficient to remove the session ID for that RUDP session.

If you want to change the SS7 session set to which a session belongs, you have to remove the entire session first. This is intended to preserve connection and recovery logic.

Examples

The following example sets up two sessions on a Cisco 2611 and creates session set 2:

```
ss7 session-0 address 172.16.1.0 7000 172.16.0.0 7000 session-set 2
ss7 session-1 address 172.17.1.0 7002 172.16.0.0 7001 session-set 2
```

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 cumack_t	Sets the cumulative acknowledgment timer.
ss7 session k_pt	Sets the null segment (keepalive) timer.
ss7 session m_cumack	Sets the maximum number of segments that can be received before the RUDP sends 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_retrans	Sets the maximum number of times that the RUDP attempts to resend a segment before declaring the connection invalid.
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 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 reset to the default, 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	Interval, in milliseconds, that the RUDP waits before it sends an acknowledgment after receiving a segment. Range is from 100 to 65535. The value should be less than the value configured for the retransmission timer by using the **ss7 session-***session number* **retrans_t** command.

Defaults

300 ms

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 ms 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 reset to the default, 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	Interval, 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. Default is 2000.

Defaults

2000 ms

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 ms 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 reset to the default, 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	Maximum number of segments that can be received before the Reliable User Datagram Protocol (RUDP) sends an acknowledgment. Range is from 0 to 255. Default is 3.

Defaults

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 reset to the default, 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	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. Range is from 0 to 255. Default is 3.

Defaults

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 reset to the default, 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	Maximum number of segments that the remote (Cisco IOS software) end can send before receiving an acknowledgment. Range is from 1 to 64. Default is 32.

Defaults

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 reset to the default, 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	Maximum number of times that the RRUDP attempts to resend a segment before declaring the connection broken. Range is from 0 to 255. Default is 2.

Defaults

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 reset to the default, 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	Amount of time, in milliseconds, that the RUDP waits to receive an acknowledgment for a segment. Range is from 100 to 65535. Default is 600.

Defaults

600 ms

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 ms 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
show ss7	Displays the SS7 configuration.
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 set

Note Effective with Cisco IOS Release 12.2(15)T, the the ss7 set command replaces the **ss7 set failover-timer** command.

To independently select failover-timer values for each session set and 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 (MGC) to indicate that the Cisco Signaling Link Terminal (SLT) should switch traffic to the standby session, use the **ss7 set** command in global configuration mode. To restore the restore the failover timer to its default value of 5, use the **no** form of this command.

ss7 set [**session-set** *session-id*] **failover-timer** *ft-value*

no ss7 set [**session-set** *session-id*] **failover-timer**

Syntax Description

session-set *session-id*	(Optional) Selects failover timer values for each SS7 session set. Valid values are from 1 to 5. Default is 0.
failover-timer *ft-value*	Time, in seconds, that the Session Manager waits for a session to recover. Valid values range from 1 to 10. Default is 5.

Defaults

The failover timer is not set.

Command Modes

Global configuration

Command History

Release	Modification
12.2(15)T	This command was introduced. This command replaces the **ss7 set failover-timer** command.

Usage Guidelines

The **failover-timer** keyword and the *ft-value* argument specify the number of seconds that the Session Manager waits for the active session to recover or for the standby MGC to indicate that the SLT should switch traffic to the standby session and to make that session the active session. If the failover timer expires without recovery of the original session or if the system fails to get an active message from the standby MGC, the signaling links are taken out of service.

The **no** form of this command restores the failover timer to its default value of 5. Omitting the optional **session-set** keyword implicitly selects SS7 session set 0, which is the default.

Examples

The following example sets the failover timer to four seconds without using the **session-set** option:

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

The following example sets the failover timer to 10 seconds and sets the SS7 session set value to 5:

```
ss7 set session-set 5 failover-timer 10
```

Related Commands

Command	Description
ss7 session	Creates a Reliable User Datagram Protocol (RUDP) session and explicitly adds an RUDP session to a Signaling System 7 (SS7) session set.
ss7 set failover timer	Specifies the amount of time that the Session Manager waits for the session to recover before declaring the session inactive.

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 reset ti the default, 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. Range is from 1 to 10. Default is 3.

Defaults

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 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 4 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 name

To specify the name that is to be sent as caller ID information and to enable caller ID, use the **station name** command in voice-port configuration mode 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, use the **no** form of this command.

station name *name*

no station name *name*

Syntax Description

name	Station name. Must be a string of 1 to 15 characters.

Defaults

The default is no station name.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

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 Central Office (CO), to suit situations in which 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 is not 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 name**, **station number**, or a **caller-id alerting** command is configured on the voice port, caller ID is automatically enabled, and the **caller-id enable** command is not necessary.

Examples

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

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

Related Commands

Command	Description
caller-id enable	Enables caller ID operation.
station number	Enables caller ID operation and specifies the number sent from the originating station or network FXO port for caller ID purposes.

station number

To specify the telephone or extension number that is to be sent as caller ID information and to enable caller ID, use the **station number** command in voice-port configuration mode 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 number, use the **no** form of this command.

station number *number*

no station number *number*

Syntax Description

number	Station number. Must be a string of 1 to 15 characters.

Defaults

The default is no station number.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(2)XH	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.

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 Central Office (CO), to suit situations in which 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.

Within the network, if an originating station does not include configured number information, Cisco IOS software determines the number by using reverse dial-peer search.

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

If the **station name**, **station number**, or a **caller-id alerting** command is configured on the voice port, caller ID is automatically enabled, and the **caller-id enable** command is not necessary.

Examples

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

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

Related Commands

Command	Description
caller-id enable	Enables caller ID operation.
station name	Enables caller ID operation and specifies the name sent from the originating station or network FXO port for caller ID purposes.

subaddress

To configure a subaddress for a POTS port, use the **subaddress** command in dial-peer voice configuration mode. To disable the subaddress, use the **no** form of this command.

subaddress *number*

no subaddress *number*

Syntax Description

number	Actual subaddress of the POTS port.

Defaults

No subaddress is available for a POTS port.

Command Modes

Dial-peer voice configuration

Command History

Release	Modification
12.2(8)T	This command was introduced on the Cisco 803, Cisco 804, and Cisco 813.

Usage Guidelines

You can use this command for any dial-peer voice POTS port. You can configure only one subaddress for each of the POTS ports. The latest entered subaddress on the dial-peer voice port is stored. To check the status of the subaddress configuration, use the **show running-config** command.

Examples

The following examples show that a subaddress of 20 has been set for POTS port 1 and that a subaddress of 10 has been set for POTS port 2:

```
dial-peer voice 1 pots
 destination-pattern 5555555
 port 1
 no call-waiting
 ring 0
 volume 4
 caller-number 1111111 ring 3
 caller-number 2222222 ring 1
 caller-number 3333333 ring 1
 subaddress 20

dial-peer voice 2 pots
 destination-pattern 4444444
 port 2
 no call-waiting
 ring 0
 volume 2
 caller-number 6666666 ring 2
 caller-number 7777777 ring 3
 subaddress 10
```

subcell-mux

To enable ATM adaption layer 2 (AAL2) common part sublayer (CPS) subcell multiplexing on a Cisco router, use the **subcell-mux** command in voice-service configuration mode. To reset to the default, use the **no** form of this command.

subcell-mux *time*

no subcell-mux *time*

Syntax Description

time	Timer value, in milliseconds. Range is from 5 to 1000 (1 second). Default is 10. The *time* argument is implemented for Cisco 3600 routers.

Defaults

10 ms
Subcell multiplexing is off

Command Modes

Voice-service configuration

Command History

Release	Modification
12.1(1)XA	This command was introduced on the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.2(2)XB	The *time* argument was implemented on the Cisco 3660.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

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

Examples

The following example sets AAL2 CPS subcell multiplexing to 15 ms:

```
Router(conf-voi-serv-sess)# subcell-mux 15
```

Related Commands

Command	Description
voice-service	Specifies the voice encapsulation type and enters voice-service configuration mode.

supervisory answer dualtone

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

supervisory answer dualtone [**sensitivity** {**high** | **medium** | **low**}]

no supervisory answer dualtone

Syntax Description

sensitivity	(Optional) Specific detection sensitivity for answer supervision.
high	Increased level of detection sensitivity.
medium	Default level of detection sensitivity.
low	Decreased level of detection sensitivity.

Defaults

Answer supervision is not enabled on voice ports.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.2(2)T	This command was introduced on the following platforms: Cisco 1750, Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.

Usage Guidelines

This command configures the FXO voice port to detect voice, fax, and modem traffic when calls are answered. If answer supervision is enabled, calls are not recorded as connected until answer supervision is triggered.

This command enables a ring-no-answer timeout that drops calls after a specified period of ringback. The period of ringback can be configured using the **timeouts ringing** command.

This command automatically enables disconnect supervision in the preconnect mode on the voice port if disconnect supervision is not already enabled with the **supervisory disconnect dualtone** command.

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

If false answering is detected, decrease the sensitivity setting. If answering detection is failing, increase the sensitivity setting.

Examples

The following example enables answer supervision on voice port 1/5:

```
voice-port 1/5
 supervisory answer dualtone
```

The following example enables answer supervision on voice port 0/1/1:

```
voice-port 0/1/1
 supervisory answer dualtone
```

Related Commands

Command	Description
supervisory custom-cptone	Associates a class of custom call-progress tones with a voice port.
supervisory disconnect dualtone	Enables disconnect supervision on an FXO voice port.
timeouts ringing	Specifies the time that the calling voice port allows ringing to continue if a call is not answered.
voice class custom-cptone	Creates a voice class for defining custom call-progress tones.
voice class dualtone-detect-params	Modifies the frequency, power, and cadence tolerances of call-progress tones.

supervisory custom-cptone

To associate a class of custom call-progress tones with a voice port, use the **supervisory custom-cptone** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

supervisory custom-cptone *cptone-name*

no supervisory custom-cptone

Syntax Description

cptone-name	Descriptive identifier of the class of custom call-progress tones to be detected by a voice port. This name must match the *cptone-name* of a class of tones defined by the **voice class custom-cptone** command.

Defaults

U.S. standard call-progress tones are associated with a voice port.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(5)XM	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)T	This command was implemented on the Cisco 1750.

Usage Guidelines

This command associates a class of custom call-progress tones, defined by the **voice class custom-cptone** command, with a voice port.

You can associate the same custom call-progress tones to multiple voice ports.

You can associate only one class of custom call-progress tones with a voice port. If you associate a second class of custom call-progress tones with a voice port, the second class of custom tones replaces the one previously assigned.

This command is applicable to analog Foreign Exchange Office (FXO) voice ports with loop-start signaling.

Examples

The following example associates the class of custom call-progress tones named country-x with voice ports 1/4 and 1/5:

```
voice-port 1/4
 supervisory custom-cptone country-x
 exit
voice-port 1/5
 supervisory custom-cptone country-x
 exit
```

Related Commands

Command	Description
dualtone	Defines a call-progress tone to be detected.
supervisory answer dualtone	Enables answer supervision on an FXO voice port.
supervisory disconnect dualtone	Enables disconnect supervision on an FXO voice port.
voice class custom-cptone	Creates a voice class for defining custom call-progress tones.

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 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 the Cisco MC3810.

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 ms. 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 interprets those tones as a disconnect indication and closes the window.

Examples

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

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

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

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

supervisory disconnect anytone

To configure a Foreign Exchange Office (FXO) voice port to go on-hook if the router detects any tone from a PBX or the PSTN before an outgoing call is answered, use the **supervisory disconnect anytone** command in voice-port configuration mode. To disable the supervisory disconnect function, 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(5)XM	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 series.
12.2(2)T	This command was integrated into Cisco IOS Release 12.2(2)T and implemented on the Cisco 1750.

Usage Guidelines

Use this command to provide disconnect if the PBX or PSTN does not provide a supervisory tone. Examples of tones that trigger a disconnect include busy tone, fast busy tone, and dial tone.

This command is enabled only during call setup (before the call is answered).

You must enable echo cancellation; otherwise, ringback tone from the router can trigger a disconnect.

This command replaces the **no supervisory disconnect signal** command. If you enter this command, the supervisory disconnect anytone feature is enabled, and the message `supervisory disconnect anytone` is displayed when **show** commands are entered.

If you enter either the **supervisory disconnect anytone** command or the **no supervisory disconnect signal** command, answer supervision is automatically disabled.

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
supervisory answer dualtone	Enables answer supervision on an FXO voice port.
supervisory disconnect dualtone	Enables disconnect supervision on an FXO voice port.
timeouts call-disconnect	Specifies the timeout value for releasing an FXO voice port when an incoming call is not answered.

supervisory disconnect dualtone

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

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

no supervisory disconnect dualtone

Syntax Description

mid-call	Disconnect supervision operates throughout the duration of the call.
pre-connect	Disconnect supervision operates during call setup and stops when the called telephone goes off-hook.

Defaults

Disconnect supervision is not enabled on voice ports.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(5)XM	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)T	This command was implemented on the Cisco 1750.

Usage Guidelines

This command configures an FXO voice port to disconnect calls when the router detects call-progress tones from a PBX or the PSTN. Disconnection occurs after the wait-release time specified on the voice port.

Disconnect supervision is automatically enabled in the preconnect mode on the voice port if the **supervisory answer dualtone** command is entered.

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

Examples

The following example specifies tone detection during the entire call duration:

```
voice-port 1/5
 supervisory disconnect dualtone mid-call
 exit
```

The following example specifies tone detection only during call setup:

```
voice-port 0/1/1
 supervisory disconnect dualtone pre-connect
 exit
```

Related Commands

Command	Description
supervisory answer dualtone	Enables answer supervision on an FXO voice port.
supervisory custom-cptone	Associates a class of custom call-progress tones with a voice port.
timeouts call-disconnect	Specifies the timeout value for releasing an FXO voice port when an incoming call is not answered.
timeouts wait-release	Specifies the timeout value for releasing a voice port when an outgoing call is not answered.
voice class dualtone-detect-params	Modifies the frequency, power, and cadence tolerances of call-progress tones.

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	Tone detection operates throughout the duration of a call.
pre-connect	Tone detection operates during call setup and stops when the called telephone goes off-hook.
tag	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. Range is from 1 to 10000.

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 series, Cisco 3600 series, and Cisco MC3810.

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.

supervisory dualtone-detect-params

To associate a class of modified tone-detection tolerance limits with a voice port, use the **supervisory dualtone-detect-params** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

supervisory dualtone-detect-params *tag*

no supervisory dualtone-detect-params

Syntax Description

tag	Tag number of the set of modified tone-detection tolerance limits to be associated with the voice port. The tag number must match the tag number of a voice class configured by the **voice class dualtone-detect-params** command. Range is from 1 to 10000.

Defaults

The default tone-detection tolerance limits are associated with voice ports.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(5)XM	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.2(2)T	This command was implemented on the Cisco 1750.

Usage Guidelines

This command associates a specific set of modified tone-detection tolerance limits, defined by the **voice class dualtone-detect-params** command, with a voice port.

You can associate the same class of modified tone-detection tolerance limits to multiple voice ports.

You can associate only one class of modified tone-detection tolerance limits to a voice port. If you associate a second class of modified tone-detection tolerance limits with a voice port, the second class replaces the one previously assigned.

This command is applicable to analog Foreign Exchange Office (FXO) voice ports with loop-start signaling.

Examples

The following example associates the class of modified tone-detection tolerance limits that has tag 70 with voice ports 1/5 and 1/6.

```
voice-port 1/5
 supervisory dualtone-detect-params 70
 exit
voice-port 1/6
 supervisory dualtone-detect-params 70
 exit
```

The following example restores the default tone-detection parameters to voice port 1/5.

```
voice-port 1/5
 no supervisory dualtone-detect-params
 exit
```

Related Commands

Command	Description
supervisory answer dualtone	Enables answer supervision on an FXO voice port.
supervisory disconnect dualtone	Enables disconnect supervision on an FXO voice port.
voice class dualtone-detect-params	Creates a voice class for call-progress tone-detection tolerance parameters.

suppress

To suppress accounting for a specific call leg, use the **suppress** command in gateway accounting AAA configuration mode. To reenable accounting for that leg, use the **no** form of this command.

suppress [**pots** | **rotary** | **voip**]

no suppress [**pots** | **rotary** | **voip**]

Syntax Description

pots	(Optional) POTS call leg.
rotary	(Optional) Rotary dial peer.
voip	(Optional) VoIP call leg.

Defaults

Accounting is enabled.

Command Modes

Gateway accounting AAA configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.

Usage Guidelines

Use this command to turn off accounting for a specific call leg.

If both incoming and outgoing call legs are of the same type, no accounting packets are generated.

Use the **rotary** keyword to suppress excess start and stop accounting records. This causes only one pair of records to be generated for every connection attempt through a dial peer.

Examples

The following example suppresses accounting for the POTS call leg.

```
suppress pots
```

Related Commands

Command	Description
debug suppress rotary	Displays connection attempt statistics.
gw-accounting aaa	Enables VoIP gateway accounting.

suspend-resume (SIP)

To enable SIP Suspend and Resume functionality, use the **suspend-resume** command in SIP user agent configuration mode. To disable SIP Suspend and Resume functionality, use the **no** form of this command.

suspend-resume

no suspend-resume

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

SIP user agent configuration

Command History

Release	Modification
12.2(15)T	This command was introduced.

Usage Guidelines

Session Initiation Protocol (SIP) gateways are now enabled to use Suspend and Resume. Suspend and Resume are basic functions of ISDN and ISDN User Part (ISUP) signaling procedures. A Suspend message temporarily halts communication (call hold), and a Resume message is received after a Suspend message and continues the communication.

Examples

The following example disables Suspend and Resume functionality:

```
Router(config)# sip-ua
Router(config-sip-ua)# no suspend-resume
```

Related Commands

Command	Description
show sip-ua status	Displays SIP UA status.
sip-ua	Enables the SIP user-agent configuration commands.