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.

Foreign Exchange Office (FXO) and Foreign Exchange Station (FXS) Voice Ports

signal {**groundstart** | **loopstart** [**live-feed**]}

no signal {**groundstart** | **loopstart**}

Ear and mouth (E&M) Voice Ports

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

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

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** | **kp-0-npa-nxx-xxxx-st-kp-yyy-yyy-yyyy-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** | **kp-0-npa-nxx-xxxx-st-kp-yyy-yyy-yyyy-st**} | **groundstart** | **loopstart**}

Syntax Description		
	groundstart	Specifies the use of groundstart signaling. Used for FXO and FXS interfaces. Groundstart signaling allows both sides of a connection to place a call and to hang up.
		Note The CAMA version of this keyword is **groundstart**. Both forms operate identically.
	loopstart	Specifies the use of loop start signaling. Used for FXO and FXS interfaces. With loopstart signaling, only one side of a connection can hang up. This is the default setting for FXO and FXS voice ports.
		Note The CAMA version of this keyword is **loopstart**. Both forms operate identically.
	live-feed	(Optional) Enables an MOH audio stream from a live feed to be directly connected to the router through an FXO port.
	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 dual tone multifrequency (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.
	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.
	lmr	Specifies the use of Land Mobile Radio signaling.

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 DTMF digits. Used for E&M tie trunk interfaces. This is the default setting for E&M voice ports.
cama	Selects and configures the port for 911 calls.
kp-0-npa-nxx-xxxx-st	10-digit transmission. The E.164 number is fully transmitted.
kp-0-npa-nxx-xxxx-st-kp-yyy-yyy-yyyy-st	Supports CAMA Signaling with ANI/Pseudo ANI (PANI).
kp-0-nxx-xxxx-st	7-digit automatic number identification (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 Numbering Plan Digit (NPD) digit in the lookup table, or when the calling number is fewer than ten digits in length. (NPA digits are not available.)
kp-npd-nxx-xxxx-st	8-digit ANI transmission, where the NPD is a single multifrequency (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

05550100 = (415) 555-0100, 15550100 = (510) 555-0100, and so on. NPD range is from 0 to 3. |

Command Default   FXO and FXS interfaces: **loopstart**
E&M interfaces: **wink-start**
CAMA interfaces: **loopstart**

Command Modes   Voice-port configuration (config-voiceport)

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.
12.3(4)XD	The **lmr** keyword was added.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(9)T	The **kp-0-npa-nxx-xxxx-st-kp-yyy-yyy-yyyy-st** keyword was added to support CAMA Signaling with ANI/Pseudo ANI (PANI).
12.4(11)XJ	The **live-feed** keyword was added.
12.4(15)T	The **live-feed** keyword was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines   This command applies to analog voice ports only. A voice port must be shut down and then activated before the configured values take effect.

For an E&M voice port, this command changes only the signal value for the selected voice port.

For an FXO or FXS voice port, this command changes the signal value for both voice ports on a voice port module (VPM). 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, see the installation documentation that came with your voice network module.

Some PBXs miss initial digits if the E&M voice port is configured for immediate start signaling. Immediate start signaling should be used for dial pulse outpulsing only and only on circuits for which the far end is configured to accept digits within a few milliseconds of seizure. Delay dial signaling, which is intended for use on trunks and not lines, relies on the far end to return an off-hook indication on its M-lead as soon as the circuit is seized. When a receiver is attached, the far end removes the off-hook indication to indicate that it is ready to receive digits. Delay dial must be configured on both ends to work properly. 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 Physical Service Access Point (PSAP) location.

The following four types of ANI transmittal schemes are based on the actual number of digits transmitted toward the E911 tandem. In each instance, the actual calling number is proceeded with a key pulse (KP) followed by an information (I) field or a NPD, which is then followed by the ANI calling number, and finally is followed by a start pulse (ST), STP, ST2P, or ST3P, depending on the trunk group type in the PSTN and the traffic mix carried.

The information field is one or two digits, depending on how the circuit was ordered originally. For one-digit information fields, a value of 0 indicates that the calling number is available. A value of 1 indicates that the calling number is not available. A value of 2 indicates an ANI failure. For a complete list of values for two-digit information fields, see *SR-2275: Telcordia Notes on the Networks* at www.telcordia.com.

- 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 415-555-0122 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.

- 20-digit transmission (**kp-0-npa-nxx-xxxx-st-kp-yyy-yyy-yyyy-st**):

  Twenty digits support (two 10 digit numbers) on FGD-OS in the following format, KP+II+10 digit ANI+ST+KP+7/10 digit PANI+ ST

- 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 out-pulsing a ten-digit ANI and the 911 call it receives has a seven-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 groundstart 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 groundstart
```

The following example configures a ten-digit ANI transmission:

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

The following example configures 20-digit CAMA Signaling with ANI/Pseudo ANI:

```
Router(config-voiceport)# signal cama KP-0-NPA-NXX-XXXX-ST-KP-YYY-YYY-YYYY-ST
```

Related Commands

Command	Description
ani mapping	Preprograms the NPA, or area code, into a single MF digit.

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.

Command Default   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 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 {*seconds* | **disabled**}

no signal keepalive {*seconds* | **disabled**}

Syntax Description

seconds	Keepalive signaling packet interval, in seconds. Range is from 1 to 65535. Default is 5 seconds.
disabled	Specifies that no keepalive signals are sent.

Command Default   *seconds*: 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.
12.3(7)T	The **disabled** keyword was added.

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 using the **voice-class permanent** (dial peer) command.

To avoid sending keepalive signals to a multicasting network with no specified destination, we recommend that you use the **disabled** keyword when configuring this command for use in networks that use connection trunk connections and multicasting.

Examples   The following example shows the keepalive signaling interval set 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.

Command Default

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.

Command Default

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* [**resume-voice** [*milliseconds*]]

> **no signal timing idle suppress-voice** *seconds* [**resume-voice** [*milliseconds*]]

Syntax Description

seconds	Duration of the idle state, in seconds, before the voice traffic is stopped. Range is from 0 to 65535.
resume-voice	(Optional) Sets a timer that controls the delay between when trunk activity is detected and when active packetization of voice resumes.
milliseconds	(Optional) Duration of the delay, in milliseconds (ms), for the resume-voice timer. Range is from 40 to 5000. Default is 500 ms.

Command Default

No signal timing idle suppress-voice timer is configured.

Command Modes

Voice-class configuration (config-voice-class)

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810 platform.
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.
12.4(15)T10	This command was modified to add the **resume-voice** *milliseconds* option.

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.

In some circumstances, the default delay of 500 ms between the detection of incoming seizure and the opening of the audio path may cause a timing issue.

If, during this delay of 500 ms, the near-end originating PBX has already received the acknowledgement from the far-end PBX to begin playing out digits and the audio path is not yet open, the first Dual Tone Multi-Frequency (DTMF) digit might be lost over the permanent trunk.

This loss of the first DTMF digit can occur if a Cisco voice gateway has the following trunk conditioning setting:

```
!
voice class permanent 1
signal pattern idle transmit 0000
signal pattern idle receive 0000
signal pattern oos transmit 1111
signal pattern oos receive 1111
signal timing idle suppress-voice 10

!
```

The **resume-voice** *milliseconds* option has been added in Release 12.4(15)T10 to modify the delay timer and reduce the wait time. We recommend that you specify a delay of less than 500 ms to avoid the loss of any digits due to the possible discrepancy between the detection of incoming seizure and the opening of the audio path.

The output of the **show voice trunk-conditioning supervisory** command has been modified in Release 12.4(15)T10 to report values for the **suppress-voice** and **resume-voice** keywords (of the **signal timing idle suppress-voice** command) as the "idle = *seconds*" and "idle_off = *milliseconds*" fields, respectively.

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.
show voice trunk-conditioning supervisory	Displays the status of trunk supervision and configuration parameters for a voice port.
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.

Command Default   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.

Command Default 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
```

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.

Related Commands appears to the left of the table header.

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.	

Command Default   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.	

Command Default 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.

Command Default

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.	

Command Default   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.

Command Default 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

Command	Description
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.

Command Default   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

Command	Description
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	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 handling of arbitrary or unknown signaling protocols.
	For an AVM: It is not possible to provide "transparent" behavior without interpreting the signaling information to read and write the correct state to the analog hardware. This option is mapped to be equal to **cas**.

Command Default   **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 for VoFR dial peer 200:

```
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.

Command Default   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.

Command Default   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-header

To specify the Session Initiation Protocol (SIP) header to be sent to the peer call leg, use the **sip-header** command in voice class configuration mode. To disable the configuration, use the **no** form of this command.

sip-header {**sip-req-uri** | *header-name*}

no sip-header {**sip-req-uri** | *header-name*}

Syntax Description

sip-req-uri	Configures Cisco Unified Border Element (UBE) to send a SIP request Uniform Resource Identifier (URI) to the peer call leg.
header-name	Name of the header to be sent to the peer call leg.

Command Default   SIP header is not sent to the peer call leg.

Command Modes   Voice class configuration (config-class)

Command History

Release	Modification
15.1(3)T	This command was introduced.

Usage Guidelines   Use the **sip-header** command to configure Cisco UBE to pass the unsupported parameters present in a mandatory header from one peer call leg to another of a Cisco UBE.

Examples   The following example shows how to configure Cisco UBE to send a "From" header to the peer call leg:

```
Router(config)# voice class sip-copylist 2
Router(config-class)# sip-header From
```

Related Commands

Command	Description
voice class sip-copylist	Configures a list of entities to be sent to a peer call leg and enters voice class 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:***ipv4-address* | **ipv6:**[*ipv6-address*][**:***port-num*]}

> **no sip-server**

Syntax Description

dns:	Sets the global SIP server interface to a Domain Name System (DNS) hostname. If you do not specify a hostname, the default DNS defined by the **ip name-server** command is used.
host-name	(Optional) Valid DNS hostname in the following format: name.gateway.xyz.
ipv4:*ipv4-address*	Sets the global SIP server interface to an IPv4 address. A valid IPv4 address takes the following format: xxx.xxx.xxx.xxx.
ipv6:[*ipv6-address*]	Sets the global SIP server interface to an IPv6 address. You must enter brackets around the IPv6 address.
:*port-num*	(Optional) Port number for the SIP server.

Command Default   No network address is configured.

Command Modes   SIP user-agent configuration (conf-serv-sip)

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 was not included in this release.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. This command was implemented on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.4(22)T	Support for IPv6 was added.

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 a Cisco SIP proxy server (SPS) is present in the network. With an SPS, you can configure the SIP server option and have the interested dial peers use the SPS by default.

To reset this command to a null value, use the **default** command.

To configure an IPv6 address, the user must enter brackets [ ] around the IPv6 address.

Examples

The following example, beginning in global configuration mode, sets the global SIP server interface to the DNS hostname "3660-2.sip.com." If you also use the **session target sip server** command, you need not set the DNS hostname 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 IPv4 address:

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

The following example sets the global SIP server interface to an IPv6 address. Note that brackets were entered around the IPv6 address:

```
sip-ua
 sip-server ipv6:[2001:0DB8:0:0:8:800:200C:417A]
```

Related Commands

Command	Description
default	Enables a default aggregation cache.
ip name-server	Specifies the address of one or more name servers to use for name and address resolution.
session target (VoIP dial peer)	Specifies a network-specific address for a dial peer.
session target sip-server	Instructs the dial peer session target to use the global SIP server.
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.

Command Default If this command is not enabled, no SIP user-agent configuration commands can be entered.

Command Modes Global configuration (config)

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.
15.1(2)T	The **connection-reuse** SIP user-agent configuration mode command was added.

Usage Guidelines Use this command to enter SIP user-agent configuration mode. Table 230 lists the SIP user-agent configuration mode commands.

Table 230 SIP User-Agent Configuration Mode Commands

Command	Description
connection-reuse	Uses the listener port for sending requests over the User Datagram Protocol (UDP).
exit	Exits SIP user-agent configuration mode.
inband-alerting	This command is no longer supported as of Cisco IOS Release 12.2 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.

Table 230   SIP User-Agent Configuration Mode Commands (continued)

Command	Description
retry	Configures the SIP signaling timers for retry attempts.
sip-server	Configures a SIP server interface.
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, shows how to enter SIP user-agent configuration mode, configure the SIP user agent, and then return to global configuration mode:

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

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.

Command Default   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.

soft-offhook

To enable stepped off-hook resistance during seizure, use the **soft-offhook** command in voice-port (FXO) configuration mode. To disable this command, use the **no** form of this command.

soft-offhook

no soft-offhook

Syntax Description   This command has no arguments or keywords.

Command Default   This command is disabled by default, which means there is no stepped off-hook resistance during seizure.

Command Modes   Voice-port (FXO) configuration (config-voiceport)

Command History

Release	Modification
12.4(3f) 12.4(4)T4	This command was introduced.

Usage Guidelines   An off-hook indication into a far-end ringing cadence ON condition can occur during glare conditions (outgoing seizure occurring at the same time as an incoming ring). This condition can also occur when the interface configuration includes the **connection plar-opx** command. If the **connection plar-opx** command is not configured, the FXO software waits for a ringing cadence to transition from ON to OFF prior to transitioning to the off-hook condition. (Glare can be minimized by configuring ground-start signaling.)

When the **soft-offhook** command is entered, the FXO hookswitch off-hook resistance is initially set to a midresistance value for outgoing or incoming seizure. This resistance limits the ringing current that occurs during seizure into ringing signals prior to far-end ring-trip. When ringing is no longer detected, hookswitch resistance is returned to its normal lower value. This prevents damage to the FXO line interface that may occur in locations with short loops and conventional ringing sources with low output impedance ringing sources that have the potential to deliver high current.

The **soft-offhook** command applies to the following FXO interface cards (which use the 3050i chipset):

- EM-HDA-3FXS/4FXO (EVM-HD-8FXS/DID, FXO ports only)
- EM-HDA-6FXO (on EVM-HD-8FXS/DID)
- EM2-HDA-4FXO (NM-HDA-4FXS network module only)
- VIC2-4FXO, VIC2-2FXO

Examples   The following example shows a sample configuration session to enable stepped off-hook resistance during seizure on voice port 1/0/0 on a Cisco 3725 router:

```
Router# configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.

Router(config)# voice-port 1/0/0
Router(config-voiceport)# soft-offhook
Router(config-voiceport)# shutdown
Router(config-voiceport)#
Nov  3 11:08:53.313 EST: %LINK-3-UPDOWN: Interface Foreign Exchange Office 1/0/0, changed
state to Administrative Shutdown

Router(config-voiceport)# no shutdown
Router(config-voiceport)#
Nov  3 11:08:58.290 EST: %LINK-3-UPDOWN: Interface Foreign Exchange Office 1/0/0, changed
state to up

Router(config-voiceport)# ^Z
Router#
Nov  3 11:09:01.086 EST: %SYS-5-CONFIG_I: Configured from console by console

Router#
```

Related Commands

Command	Description
connection plar-opx	Specifies the connection mode for a voice port as PLAR-OPX.
voice-port	Enters voice-port configuration mode.

source carrier-id

To configure debug filtering for the source carrier ID, use the **source carrier-id** command in call filter match list configuration mode. To disable, use the **no** form of this command.

source carrier-id *string*

no source carrier-id *string*

Syntax Description

string	Alphanumeric identifier for the carrier ID.

Command Default   No default behavior or values

Command Modes   Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples   The following example shows the voice call debug filter set to match source carrier ID 4321:

```
call filter match-list 1 voice
 source carrier-id 4321
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
show call filter match-list	Display call filter match lists.
source trunk-group-label	Configure debug filtering for a source trunk group.
target carrier-id	Configure debug filtering for the target carrier ID.
target trunk-group-label	Configure debug filtering for a target trunk group.

source trunk-group-label

To configure debug filtering for a source trunk group, use the **source trunk-group-label** command in call filter match list configuration mode. To disable, use the **no** form of this command.

source trunk-group-label *group_number*

no source trunk-group-label *group_number*

Syntax Description

group_number	A value from 0 to 23 that identifies the trunk group.

Command Default No default behavior or values

Command Modes Call filter match list configuration

Command History

Release	Modification
12.3(4)T	This command was introduced.

Examples The following example shows the voice call debug filter set to match source trunk group 21:

```
call filter match-list 1 voice
 source trunk-group-label 21
```

Related Commands

Command	Description
call filter match-list voice	Create a call filter match list for debugging voice calls.
debug condition match-list	Run a filtered debug on a voice call.
show call filter match-list	Display call filter match lists.
source carrier-id	Configure debug filtering for the source carrier ID.
target carrier-id	Configure debug filtering for the target carrier ID.
target trunk-group-label	Configure debug filtering for a target trunk group.

speed dial

To designate a range of digits for SCCP telephony control (STC) application feature speed-dial codes, use the **speed dial** command in STC application feature speed-dial configuration mode. To return the range to its default, use the **no** form of this command.

speed dial from *digit* **to** *digit*

no speed dial

Syntax Description		
from *digit*	Starting number for the range of speed-dial codes. Range is 0 to 9 for one-digit codes; 00 to 99 for two-digit codes. Default is 1 for one-digit codes; 01 for two-digit codes.	
	Note Range depends on the number of digits set with the **digit** command.	
to *digit*	Ending number for the range of speed-dial codes. Range is 0 to 9 for one-digit codes; 00 to 99 for two-digit codes. Default is 9 for one-digit codes; 99 for two-digit codes.	
	Note Range depends on the number of digits set with the **digit** command.	

Command Default   The default speed-dial codes are 1 to 9 for one-digit codes; 01 to 99 for two-digit codes.

Command Modes   STC application feature speed-dial configuration

Command History	Release	Modification
	12.4(2)T	This command was introduced.
	12.4(6)T	The *digit* argument was modified to allow two-digit codes.

Usage Guidelines   This command is used with the STC application, which enables features on analog FXS endpoints that use Skinny Client Control Protocol (SCCP) for call control.

Use this command to set the range of speed-dial codes only if you want to change the range from its default. The **digit** command determines whether speed-dial codes are one-digit or two-digit.

A maximum of nine one-digit or 99 two-digit speed-dial codes are supported. If you set the starting number to 0, the highest number you can set for the ending number is 8 for one-digit codes, or 98 for two-digit codes.

Note that the actual telephone numbers that are speed dialed are stored on Cisco CallManager or the Cisco CallManager Express system. The speed-dial codes that you set with this command are mapped to speed-dial positions on the call-control device. For example, if you set the starting number to 2 and the ending number to 7, the system maps 2 to speed-dial 1 and maps 7 to speed-dial 6.

You can enter numbers in this command in ascending or descending order. For example, the following commands are both valid:

```
Router(stcapp-fsd)# speed dial from 2 to 7
Router(stcapp-fsd)# speed dial from 7 to 2
```

To use the speed-dial feature on a phone, dial the STC application feature speed-dial (FSD) prefix and one of the speed-dial codes that has been configured with this command (or the default if this command was not used). For example, if the FSD prefix is * (the default) and the speed-dial codes are 1 to 9 (the default), dial *3 to dial the telephone number stored with speed-dial 3.

This command resets to its default range if you modify the value of the **digit** command. For example, if you set the **digit** command to 2, then change the **digit** command back to its default of 1, the speed-dial codes are reset to 1 to 9.

If the **digit** command is set to 2 and you configure a single-digit speed-dial code, the system converts the speed-dial code to two digits. For example, if you enter the range 1 to 5 in a two-digit configuration, the system converts the speed-dial codes to 11 to 15.

If you set any of the FSD codes in this range to a value that is already in use for another FSD code, you receive a warning message. If you configure a duplicate code, the system implements the first matching feature in the order of precedence shown in the output of the **show stcapp feature codes** command.

The **show running-config** command displays nondefault FSD codes only. The **show stcapp feature codes** command displays all FSD codes.

Examples

The following example sets an FSD code prefix of two pound signs (##) and a speed-dial code range of 2 to 7. After these values are configured, a phone user presses ##2 to dial the number that is stored with speed-dial 1 on the call-control system (Cisco CallManager or Cisco CallManager Express).

```
Router(config)# stcapp feature speed-dial
Router(stcapp-fsd)# prefix ##
Router(stcapp-fsd)# speed dial from 2 to 7
Router(stcapp-fsd)# exit
```

The following example shows how the speed-dial range that is set in the example above is mapped to the speed-dial positions on the call-control system. Note that the range from 2 to 7 is mapped to speed-dial 1 to 6.

```
Router# show stcapp feature codes
.
.
.
  stcapp feature speed-dial
    prefix ##
    redial ###
    speeddial number of digit(s) 1
    voicemail ##0
    speeddial1 ##2
    speeddial2 ##3
    speeddial3 ##4
    speeddial4 ##5
    speeddial5 ##6
    speeddial6 ##7
```

The following example sets a FSD code prefix of two asterisks (**) and a speed-dial code range of 12 to 17.

```
Router(config)# stcapp feature speed-dial
Router(stcapp-fsd)# prefix **
Router(stcapp-fsd)# digit 2
Router(stcapp-fsd)# speed dial from 12 to 17
Router(stcapp-fsd)# exit
```

Related Commands

Command	Description
digit	Designates the number of digits for STC application feature speed-dial codes.
prefix (stcapp-fsd)	Designates a prefix to precede the dialing of an STC application feature speed-dial code.
redial	Designates an STC application feature speed-dial code to dial again the last number that was dialed.
show running-config	Displays current nondefault configuration settings.
show stcapp feature codes	Displays configured and default STC application feature access codes.
stcapp feature speed-dial	Enters STC application feature speed-dial configuration mode to set feature speed-dial codes.
voicemail (stcapp-fsd)	Designates an STC application feature speed-dial code to dial the voice-mail number.

srtp (dial peer)

To specify that Secure Real-Time Transport Protocol (SRTP) be used to enable secure calls for a specific VoIP dial peer, to enable fallback, and to override global SRTP configuration, use the **srtp** command in dial peer voice configuration mode. To disable secure calls, to disable fallback, and to override global SRTP configuration, use the **no** form of this command.

> **srtp** [**fallback** | **system**]

> **no srtp** [**fallback** | **system**]

Syntax Description		
fallback	(Optional) Enables specific dial-peer calls to fall back to nonsecure mode.	
system	(Optional) Enables the global SRTP configuration that was set using the **srtp** command in voice service voip configuration mode. This is the default if the **srtp** command is enabled in dial peer voice configuration mode.	

Command Default Global SRTP configuration set in voice service voip configuration mode is enabled.

Command Modes Dial peer voice configuration

Command History

Release	Modification
12.4(6)T1	This command was introduced.

Usage Guidelines You can enable secure calls using the **srtp** command either at the dial peer level, or at the global level. The **srtp** command in dial peer voice mode configures call security at the dial-peer level and takes precedence over the global **srtp** command. Use the **srtp** command in dial peer voice configuration mode to enable secure calls for a specific dial peer. Use the **no** form of this command to disable secure calls.

Use the **srtp fallback** command to enable secure calls and allow calls to fallback to nonsecure mode for a specific dial peer. This security policy applies to all calls going through the dial peer and is not configurable on a per-call basis. Using the **srtp fallback** command to configure call fallback at the dial-peer level takes precedence over the global **srtp fallback** command. The **no** form of this command disables SRTP and fallback. If you disallow fallback using the **no srtp fallback** command, a call cannot fall back to nonsecure mode.

Use the **srtp system** command to apply global level security settings to dial peers.

Examples   The following example enables secure calls and disallows fallback for a specific dial peer:

```
Router(config-dial-peer)# srtp
```

The following example enables secure calls and allows call fallback to nonsecure mode:

```
Router(config-dial-peer)# srtp fallback
```

The following example defaults call security to global level SRTP behavior:

```
Router(config-dial-peer)# srtp system
```

Related Commands

Command	Description
srtp (voice)	Enables secure calls globally in voice service voip configuration mode.
srtp fallback (voice)	Enables SRTP and fallback globally.

srtp (voice)

To specify that Secure Real-Time Transport Protocol (SRTP) be used to enable secure calls and call fallback, use the **srtp** command in voice service voip configuration mode. To disable secure calls and disallow fallback, use the **no** form of this command.

> **srtp** [**fallback**]

> **no srtp** [**fallback**]

Syntax Description

fallback	(Optional) Enables call fallback to nonsecure mode.

Command Default   Voice call security and fallback are disabled.

Command Modes   Voice service voip configuration

Command History

Release	Modification
12.4(6)T1	This command was introduced.

Usage Guidelines   Use the **srtp** command in voice service voip configuration mode to globally enable secure calls using SRTP media authentication and encryption. This security policy applies to all calls going through the gateway and is not configurable on a per-call basis. To enable secure calls for a specific dial peer, use the **srtp** command in dial peer voice configuration mode. Using the **srtp** command to configure call security at the dial-peer level takes precedence over the global **srtp** command.

Use the **srtp fallback** command to globally enable secure calls and allow calls to fall back to RTP (nonsecure) mode. This security policy applies to all calls going through the gateway and is not configurable on a per-call basis. To enable secure calls for a specific dial peer, use the **srtp** command in dial peer voice configuration mode. Using the **srtp fallback** command in dial peer voice configuration mode to configure call security takes precedence over the **srtp fallback** global command in voice service voip configuration mode. If you use the **no srtp fallback** command, fallback from SRTP to RTP (secure to nonsecure) is disallowed.

Examples   The following example enables secure calls:

```
Router(config-voi-serv)# srtp
```

The following example enables call fallback to nonsecure mode:

```
Router(config-voi-serv)# srtp fallback
```

Related Commands	Command	Description
	srtp (dial peer)	Enables secure calls on an individual dial peer.
	srtp fallback (dial peer)	Enables call fallback to RTP (nonsecure) mode on an individual dial peer.
	srtp fallback (voice)	Enables call fallback globally to RTP (nonsecure) mode.
	srtp system	Enables secure calls on a global level.

srv version

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

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

no srv version

Syntax Description		
	1	Specifies the domain-name prefix of format protocol.transport. (RFC 2052 style).
	2	Specifies the domain-name prefix of format _protocol._transport. (RFC 2782 style).

Defaults   2 (RFC 2782 style)

Command Modes   SIP UA configurationn (config-sip-ua)

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 was 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.
	Cisco IOS XE Release 2.5	This command was integrated into Cisco IOS XE Release 2.5.

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 configuring 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.