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timing delay-duration

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

timing delay-duration *time*
no timing delay-duration *time*

Syntax Description

time	Delay signal duration for delay dial signaling, in milliseconds. Range is from 100to 5000. The default is 2000.

Command Default

2000 milliseconds

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The call direction for the **timing delay-duration** command is out. This command is supported on E&M ports only.

Examples

The following example sets the delay signal duration on a voice port to 3000 milliseconds:

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

Related Commands

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

Command	Description
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing delay-start

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

timing delay-start *time*
no timing delay-start

Syntax Description

time	Minimum delay time, in milliseconds, from outgoing seizure to outdial address. Range is from 20to 2000. The default on the Cisco 3600 series is 300.

Command Default

Cisco 3600 series: 300 milliseconds

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The call direction for the **timing delay-start** command is out. It is supported on E&M ports only.

Examples

The following example sets the delay-start duration on a voice port to 250 milliseconds:

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

Related Commands

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

Command	Description
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing delay-voice tdm

To specify the delay after which voice packets are played out, use the **timing delay-voice tdm** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

timing delay-voice tdm *milliseconds*
no timing delay-voice tdm *milliseconds*

Syntax Description

milliseconds	Duration, in milliseconds, of the timing delay. Range is integers from 1 to 1500. Default is 0.

Command Default

milliseconds : 0 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
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(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

The **timing delay-voice tdm** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). To avoid voice loss at the receiving end of an LMR system, use this command to configure a delay for the voice packet equal to the sum of the durations of all the injected tones and pauses configured with the **inject tone** command and the **inject pause** command.

Examples

The following example configures a timing delay of 470 milliseconds before the voice packet is played out:

```
voice class tone-signal mytones
 inject tone 1 1950 3 150
 inject tone 2 2000 0 60
 inject pause 3 60
 inject tone 4 2175 3 150
 inject tone 5 1000 0 50
voice-port 1/0/0
 voice-class tone-signal mytones
 timing delay-voice tdm 470
```

Note that the delay of 470 milliseconds is equal to the sum of the durations of the injected tones and pauses in the tone-signal voice class.

Related Commands

Command	Description
inject pause	Specifies a pause between injected tones.

Command	Description
inject tone	Specifies a wakeup or frequency selection tone to be played out before the voice packet.

timing delay-with-integrity

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

timing delay-with-integrity *time*
no timing delay-with-integrity

Syntax Description

time	Duration of the wink pulse for the delay dial, in milliseconds. Range is from 0 to 5000. The default is 0.

Command Default

0 milliseconds

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 is supported on E&M ports only.

Examples

The following example sets the duration of the wink pulse for the delay dial to 10 milliseconds:

```
voice-port 1/0/0
 timing delay-with-integrity 10
```

Related Commands

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

Command	Description
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing dialout-delay

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

timing dialout-delay *time*
no timing dialout-delay *time*

Syntax Description

time	Dial-out delay, in milliseconds, for the sending digit or cut-through on a Foreign Exchange Office (FXO) trunk or an E&M immediate trunk. Range is from 100 to 5000. The default is 300.

Command Default

300 milliseconds

Command Modes

Voice-port configuration

Command History

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

Examples

The following example sets the dial-out delay to 350 milliseconds:

```
voice-port 1/0/0
 timing dialout-delay 350
```

Related Commands

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

Command	Description
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing dial-pulse min-delay

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

timing dial-pulse min-delay *time*
no timing dial-pulse min-delay

Syntax Description

time	Time between wink-like pulses, in milliseconds. Range is from 0to 5000. The default is 300.

Command Default

300 milliseconds

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

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

Examples

The following example sets the time between the generation of wink-like pulses on a voice port to 350 milliseconds:

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

Related Commands

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

Command	Description
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing digit

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

timing digit *time*
no timing digit

Syntax Description

time	The DTMF digit signal duration, in milliseconds. Range is 5 from 0to 100. The default is 100.

Command Default

100 milliseconds

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The call signal direction for the **timing digit** command is out. This command is supported on Foreign Exchange Office (FXO), Foreign Exchange Station (FXS), and E&M ports.

Examples

The following example sets the DTMF digit signal duration on a voice port to 50 milliseconds:

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

Related Commands

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

Command	Description
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing guard-out

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

timing guard-out *time*
no timing guard-out

Syntax Description

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

Command Default

The default is 2000 milliseconds

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

This command is supported on FXO voice ports only.

For Caller ID to work for FXO ports registered to a Cisco Unified CM, the range in milliseconds must be between 1000 to 2000.

Examples

The following example sets the timing guard-out duration on a voice port to 1000 milliseconds:

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

timing hangover

To specify the number of milliseconds of delay before the digital signal processor (DSP) tells Cisco IOS software to turn off the E-lead after the DSP detects that the voice stream has stopped, use the **timing hangover** command in voice-port configuration mode. To return to the default value, use the **no** form of this command.

timing hangover *milliseconds*
no timing hangover *milliseconds*

Syntax Description

milliseconds	The number of milliseconds for which the E-lead stays active after VAD determines that the voice stream has stopped. Valid values are 0 to 10000. The default is 250 milliseconds.

Command Default

milliseconds : 250 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.

Usage Guidelines

The **timing hangover**command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). If the voice port has been configured with the **lmr e-lead voice** command, use the **timing hangover** command to adjust the timing if the E-lead is being turned on and off too frequently.

Examples

The following example configures E-lead on voice port 1/0/1 on a Cisco 3745 to stay active for 300 milliseconds after VAD determines that the voice stream has stopped:

```
voice-port 1/0/1
 timing hangover 300
```

timing hookflash-in

To specify the maximum duration of an on-hook condition that will be interpreted as a hookflash by the Cisco IOS software, use the **timing hookflash-in** command in voice-port configuration mode. To restore the default duration for hookflash timing, use the **no** form of this command.

timing hookflash-in *milliseconds*
no timing hookflash-in

Syntax Description

milliseconds	Upper limit of the hookflash duration range, in milliseconds. • E&M voice ports--Range is 0 to 1550 milliseconds. Default is 480 milliseconds. • FXS voice ports--Range is 50 to 1550 milliseconds. Default is 1000 milliseconds.

Command Default

milliseconds : 480 milliseconds for E&M voice ports, 1000 milliseconds for FXS voice ports.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 3600 series.
12.3(7)T	Lower limit of the range for E&M voice ports was extended to 0 milliseconds.
12.3(14)T	This command was implemented on the Cisco 2800 series and Cisco 3800 series.
12.4(2)T	This command was integrated into Cisco IOS Release 12.4(2)T.

Usage Guidelines

This command is applied to E&M or Foreign Exchange Station (FXS) interfaces.

For Land Mobile Radio E&M voice ports, the **timing hookflash-in** command configures the delay between when the M-lead is raised and when voice is transmitted. Setting the hookflash duration to 0 milliseconds specifies no delay in the audio input and eliminates front-end clipping.

Analog phones connected to FXS ports use hookflash to access a second dial tone to initiate some phone features, such as transfer and conference. Hookflash is an on-hook condition of short duration that is usually generated when a phone user presses the Flash button on a phone. Cisco voice gateways measure the duration of detected on-hook conditions to determine whether they should be interpreted as hookflash or not. The duration for the on-hook conditions generated by Flash buttons on phones varies for different phone types and is interpreted by Cisco IOS software as follows:

- An on-hook condition that lasts for a time period that falls inside the hookflash duration range is considered a hookflash.
- An on-hook condition that lasts for a shorter period than the lower limit of the range is ignored.

- An on-hook condition that lasts for a longer period than the higher limit of the range is considered a disconnect.

The hookflash duration range for FXS voice ports is defined as follows:

- The lower limit of the range is set in software at 150 ms, although there is also a hardware-imposed lower limit that is typically about 20 ms, depending on platform type. An on-hook condition that lasts for a shorter time than this hardware-imposed lower limit is simply not reported to the Cisco IOS software.
- The upper limit of the range is set in software at 1000 ms by default, although this value can be changed using the **timing hookflash-in** command in voice-port configuration mode on the voice gateway. The upper limit can be set to any value from 50 to 1550 ms. For more information, see the explanations in the "Examples" section.

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

Examples

The following example sets an upper limit of 200 milliseconds for the hookflash duration range:

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

If the **timing hookflash-in** command is set to X, a value greater than 150, then any on-hook duration between 150 and X is interpreted as a hookflash. For example, if X is 1550, the hookflash duration range is 150 to 1550 ms. An on-hook signal that lasts for 1250 ms is interpreted as a hookflash, but an on-hook signal of 55 ms is ignored.

```
voice-port 1/0/0
 timing hookflash-in 1550
```

If the **timing hookflash-in** command is set to X, a value less than 150, then any on-hook duration between Y, the hardware lower limit, and X is interpreted as a hookflash. For example, if X is 65, the hookflash duration range is Y to 65 ms. An on-hook signal that lasts for 1250 ms is interpreted as a disconnect, but an on-hook signal of 55 ms is interpreted as a hookflash. (This example assumes that Y for the voice gateway is lower than 55 ms.)

```
voice-port 1/0/0
 timing hookflash-in 65
```

Related Commands

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

timing hookflash-out

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

timing hookflash-out *time*
no timing hookflash-out

Syntax Description

time	Duration of the hookflash, in milliseconds. Range is from 50 to 1550. The default is 400 milliseconds.

Command Default

400 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on Cisco 2500, Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, and Cisco MC3810.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(4)T	Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on Cisco AS5850.

Usage Guidelines

This command does not affect whether hookflash relay is enabled; hookflash relay is enabled only when the **dtmf-relay h245-signal** command is configured on the applicable VoIP dial peers. Hookflash is relayed by using an H.245-signal indication and can be sent only when an H.245 signal is available.

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

Examples

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

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

Related Commands

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

timing ignore m-lead

To ignore M-lead or voice activity detection (VAD) changes for a specified amount of time after sending the E-lead off signal, use the **timing ignore m-lead** command in voice-port configuration mode. To return to the default value, use the **no** form of this command.

timing ignore m-lead *milliseconds*
no timing ignore m-lead *milliseconds*

Syntax Description

milliseconds	The number of milliseconds following the sending of the E-lead off signal for which the M-lead and VAD changes are ignored. Valid values are 0 to 10000. The default is 0 milliseconds.

Command Default

milliseconds : 0 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(4)XD	This command was introduced.
12.3(7)T	This command was integrated into Cisco IOS Release 12.3(7)T.

Usage Guidelines

- The **timing ignore m-lead** command has an effect on an ear and mouth (E&M) voice port only if the signal type for that port is Land Mobile Radio (LMR). Use this command to reduce echo feedback on an LMR voice port. This command has an effect only if the voice port is configured for half duplex mode.

Examples

The following example configures voice port 1/0/1 on a Cisco 3745 to ignore M-lead or VAD changes for 500 milliseconds after sending the E-lead off signal:

```
voice-port 1/0/1
 timing ignore m-lead 500
```

timing interdigit

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

timing interdigit *time*
no timing interdigit *time*

Syntax Description

time	DTMF interdigit duration, in milliseconds. Range is from 50to 500. The default is 100.

Command Default

100 milliseconds

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The call signal direction for the **timing interdigit** command is out. This command is supported on Foreign Exchange Office (FXO), Foreign Exchange Station (FXS), and E&M ports.

Examples

The following example sets the DTMF interdigit duration on a voice port to 150 milliseconds:

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

Related Commands

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

Command	Description
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing opx-ringwait

To set the maximum wait time for detecting the next ring on FXO ports, use the **timing opx-ringwait**command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

timing opx-ringwait *msecs*
no timing opx-ringwait

Syntax Description

msecs	Maximum duration, in milliseconds, to wait for the next ring. Range is 2000 to 10000. Default is 6000.

Command Default

Timeout for detecting ring tones is 6000 ms (6 sec).

Command Modes

Voice-port configuration

Command History

Release	Modification
12.4(4)T	This command was introduced.

Usage Guidelines

This command prevents the voice gateway from prematurely disconnecting private line automatic ring-down (PLAR) off-premises extension (OPX) calls when the duration between ring tones from the switch is more than 6 sec. The absence of a ring tone from the switch indicates that the originating party has disconnected the call. Because some analog switches take longer than 6 sec to generate the ring tone, the voice gateway could clear the call leg while it is still ringing for a PLAR OPX call, unless the 6-sec default is changed with this command.

Examples

The following example sets the timeout for the next ring to 8 sec:

```
voice-port 2/0/10
 timing opx-ringwait 8000
```

Related Commands

Command	Description
voice-port	Enters voice-port configuration mode.
show voice port	Displays configuration information about a specific voice port.

timing percentbreak

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

timing percentbreak *percent*
no timing percentbreak

Syntax Description

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

Command Default

50 percent

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

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

Examples

The following example sets the break period percentage on a voice port to 30 percent:

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

Related Commands

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

timing pulse

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

timing pulse *pulses-per-second*
no timing pulse *pulses-per-second*

Syntax Description

pulses -per-second	Pulse dialing rate, in pulses per second. Range is from 10to 20. The default is 20.

Command Default

20 pulses per seconds

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The call signal direction for the **timing pulse** command is out. This command is supported on Foreign Exchange Office (FXO) and E&M ports only.

Examples

The following example sets the pulse dialing rate on a voice port to 15 pulses per second:

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

Related Commands

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

Command	Description
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing pulse-interdigit

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

timing pulse-interdigit *time*
no timing pulse-interdigit *time*

Syntax Description

time	Pulse dialing interdigit timing, in milliseconds. Range is from 100to 1000. The default is 500.

Command Default

500 milliseconds

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The call signal direction for the **timing pulse-interdigit** command is out. This command is supported on Foreign Exchange Office (FXO) and E&M ports only.

Examples

The following example sets the pulse-dialing interdigit timing on a voice port to 300 milliseconds:

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

Related Commands

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

Command	Description
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing sup-disconnect

To define the minimum time to ensure that an on-hook indication is intentional and not an electrical transient on the line before a supervisory disconnect occurs (based on power denial signaled by the PSTN or PBX), use the **timing sup-disconnect** command in voice-port configuration mode. To restore the default value, use the **no** form of this command.

timing sup-disconnect *milliseconds*
no timing sup-disconnect *milliseconds*

Syntax Description

milliseconds	Minimum time, in milliseconds, after detection of an on-hook indication to determine that the on-hook condition is intentional and then to hang up the POTS call leg. The range is from 50 to 1500. The default is 350.

Command Default

The default minimum time is 350 milliseconds before a supervisory disconnect occurs.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.3(12)	This command was introduced.
12.3(11)T6	This command was integrated into Cisco IOS Release 12.3(11)T6.
12.3(14)T	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(12)	This command was integrated into Cisco IOS Release 12.4(12).

Usage Guidelines

Prior to the implementation of the **timing sup-disconnect** command, analog Foreign Exchange Office (FXO) ports could not detect short disconnect signals lasting fewer than 350 ms in duration. Using this command, you can specify a wait period from 50 to 1500 ms to ensure that when an on-hook indication persists for a time that is longer than the configured value, the on-hook condition is considered intentional and a hang-up is signaled on the POTS call leg.

This timer affects only analog loop-start FXO voice ports.

Even though the **timing sup-disconnect** command can be entered under the voice port in FXO ground-start signaling, the changes in the timer setting take effect only in FXO loop-start signaling.

Examples

The following example sets the timer to wait 500 ms after detecting an on-hook signal before a supervisory disconnect occurs on the POTS call leg:

```
voice-port 1/0/0
 timing sup-disconnect 500
```

Related Commands

Command	Description
show voice port	Displays configuration information about a specific voice port.

Command	Description
voice-port	Enters voice-port configuration mode.

timing wait-wink

To set the maximum time to wait for wink signal after an outgoing seizure is sent, use the **timing wait-wink**command in voice port configuration mode. To restore the default value, use the **no** form of this command.

timing wait-wink *milliseconds*
no timing wait-wink *milliseconds*

Syntax Description

milliseconds	Maximum time to wait for wink signal after an outgoing seizure is sent. Valid entries are from 100to 6500 milliseconds (ms). Supported on ear and mouth (E&M) ports only.

Command Default

milliseconds : 550 milliseconds

Command Modes

Voice port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series routers.
11.3(1)MA	This command was implemented on Cisco MC3810 multiservice concentrators.
12.4(12)	The millisecond range was extended from 5000 to 6500.

Examples

The following example configures the maximum time to wait for wink signaling after an outgoing seizure is sent on a voice port for 300 milliseconds:

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

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port.

Command	Description
timing dialout-delay	Specifies the dial-out delay for the sending digit on a specified voice port.
timing delay-with-integrity	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-wait	Specifies the maximum wink-wait duration for a specified voice port.

timing wink-duration

To specify the timing for transmit and receive wink-signal duration for a voice port, use the **timing wink-duration**command in voice-port configuration mode. To reset to the default values, use the **no** form of this command.

timing wink-duration {*time* | **receive** *minimum maximum*}
no timing wink-duration

Syntax Description

time	Maximum transmit duration, in milliseconds (ms), for a wink-start signal. The range is from 50to 3000. The default is 200.
receive	Indicates that a range is to be specified for a received wink-start signal.
minimum	Received minimum wink length, in milliseconds. The range is from 40 to 2950. The default is 140.
maximum	Received maximum wink length, in milliseconds. The range is from 150 to 3150. The default is 290.

Command Default

Transmit wink-duration timing is set to 200 ms. The received wink-duration timing minimum is set to 140 ms and the maximum is set to 290 ms.

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series.
11.3(1)MA	This command was integrated into Cisco IOS Release 11.3(1)MA and support was added for the Cisco MC3810.
12.4(13)	This command was integrated into Cisco IOS Release 12.4(13) and the **receive** keyword and *minimum* and *maximum* arguments were added.

Usage Guidelines

The call signal direction for the **timing wink-duration** command is out. This command is supported on ear and mouth (E&M) ports only.

When wink-start signaling is used, the originating side seizes the line by going off-hook and then waits for an acknowledgment from the other end before initiating a call. The acknowledgment is a reversal of polarity (off-hook) for a timing period referred to as a wink. A wink should occur no earlier than 100 ms after the receipt of the incoming seizure signal. In addition to the signaling function, the wink start serves as an integrity check that identifies a malfunctioning trunk and allows the network to send a reorder tone to the calling party.

When you set the receive range, the minimum and maximum values of acceptable wink must provide an acceptable range of at least 50 ms. For example, entering the command **timing wink-duration receive 160 200** results in an error message.

Examples

The following example shows how to set the transmit wink-signal duration on voice port 1/0/0 to 300 ms:

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

The following example shows how to set the range for the receive wink-signal duration on voice port 1/0/0 to 160 to 210 ms:

```
voice-port 1/0/0
 timing wink-duration receive 160 210
```

Related Commands

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

timing wink-wait

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

timing wink-wait *time*
no timing wink-wait

Syntax Description

time	Maximum wink-wait duration, in milliseconds, for a wink start signal. Range is from 100to 6500. The default is 200.

Command Default

200 milliseconds

Command Modes

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on Cisco 3600 series.
11.3(1)MA	This command was supported on Cisco MC3810.
12.4(12)	The millisecond range was extended from 5000 to 6500.

Usage Guidelines

The call signal direction for the **timing wink-wait** command is out. This command is supported on ear and mouth (E&M) ports only.

Examples

The following example sets the wink-wait duration on a voice port to 300 milliseconds:

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

Related Commands

Command	Description
timeouts initial	Configures the initial digit timeout value for a specified voice port.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.
timeouts wait-release	Configures the timeout value for releasing voice ports.
timing clear-wait	Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port.
timing delay-duration	Specifies the delay signal duration for a specified voice port.
timing delay-start	Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port.

Command	Description
timing delay-with-integrity	Specifies the duration of the wink pulse for the delay dial for a specified voice port.
timing dialout-delay	Specifies the dialout delay for the sending digit on a specified voice port.
timing dial-pulse min-delay	Specifies the time between wink-like pulses for a specified voice port.
timing digit	Specifies the DTMF digit signal duration for a specified voice port.
timing interdigit	Specifies the DTMF interdigit duration for a specified voice port.
timing percentbreak	Specifies the percentage of a break period for a dialing pulse for a specified voice port.
timing pulse	Specifies the pulse dialing rate for a specified voice port.
timing pulse-interdigit	Specifies the pulse interdigit timing for a specified voice port.
timing wink-duration	Specifies the maximum wink signal duration for a specified voice port.

tls

To enable Transport Layer Security (TLS) for the Skinny Client Control Protocol (SCCP) connection between the SCCP server and the SCCP client, use the **tls**command in DSP farm profile configuration mode. To disable secure SCCP signaling, use the **no** form of this command.

tls
no tls

Syntax Description

This command has no arguments or keywords.

Command Default

Secure SCCP signaling exchange is enabled by default.

Command Modes

DSP farm profile configuration (config-dspfarm-profile #)

Command History

Release	Modification
12.4(22)YB	This command was introduced.
12.4(24)T	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

Use the **tls**command to enable secure SCCP signaling exchange. The configuration can be modified only when the dspfarm profile is shut down. To shut down the dsp farm profile, configure the **no shutdown** command.

Examples

The following example shows how to configure the **tls**command to enable TLS support for digital signal processor (DSP) farm services profile 1:

```
Router(config)# dspfarm profile 1 transcode security
Router(config-dspfarm-profile)# tls
```

Related Commands

Command	Description
dspfarm profile	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.

toggle-between-two-calls

To define a Feature Access Code (FAC) to access the Toggle Between Two Calls feature in feature mode on analog phones connected to FXS ports, use the **toggle-between-two-calls** command in STC application feature-mode call-control configuration mode. To return the code to its default, use the **no** form of this command.

toggle-between-two-calls *keypad-character*
no toggle-between-two-calls

Syntax Description

keypad-character	Character string of one to four characters that can be dialed on a telephone keypad (0-9, *, #). Default is #5.

Command Default

The default value is #5.

Command Modes

STC application feature-mode call-control configuration (config-stcapp-fmcode)

Command History

Release	Modification
15.0(1)M	This command was introduced.

Usage Guidelines

This command changes the value of the FAC for Toggle Between Two Calls from the default (#5) to the specified value.

If you attempt to configure this command with a value that is already configured for another FAC in feature mode, you receive a message. This message will not prevent you from configuring the feature code. If you configure a duplicate FAC, the system implements the first feature it matches in the order of precedence as determined by the value for each FAC (#1 to #5).

If you attempt to configure this command with a value that precludes or is precluded by another FAC in feature mode, you receive a message. If you configure a FAC to a value that precludes or is precluded by another FAC in feature mode, the system always executes the call feature with the shortest code and ignores the longer code. For example, 1 will always preclude 12 and 123. These messages will not prevent you from configuring the feature code. You must configure a new value for the precluded code in order to enable phone user access to that feature.

Examples

The following example shows how to change the value of the feature code for the Toggle Between Two Calls feature from the default (#5). With this configuration, a phone user in basic call mode presses hook flash to get the first dial tone, then dials an extension number to connect to a second call. During the second call, the user presses a hook flash to get a feature tone and then dials 55 to toggle back to the previous call party.

```
Router(config)# stcapp call-control mode feature
Router(config-stcapp-fmcode)# toggle-between-two-calls 55
Router(config-stcapp-fmcode)# exit
```

Related Commands

Command	Description
conference	Defines FAC in Feature Mode to initiate a three-party conference.
drop-last-conferee	Defines FAC in feature mode to use to drop last active call during a three-party conference.
hangup-last-active-call	Defines FAC in feature mode to drop last active call during a three-party conferencee.
transfer	Defines FAC in feature mode to connect a call to a third party that the phone user dials.

token-root-name

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

token-root-name *name*
no token-root-name

Syntax Description

name	Certificate identification name as configured with the **crypto ca identity** *name* command or the **crypto ca trusted-root** *name* command.

Command Default

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

Command Modes

Settlement configuration

Command History

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

Examples

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

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

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

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

Related Commands

Command	Description
crypto ca identity	Declares the Certificate Authority that your router should use.

Command	Description
crypto ca trusted -root	Configures the root certificate that the server uses to sign the settlement tokens.
show settlement	Displays the configuration for all settlement server transactions.

tone busytone

To enable automatic busytone generation in a basic call scenario, use the **tone busytone**command in dial peer voice configuration mode. To disable automatic busytone generation, use the **no** form of this command.

tone busytone remote-onhook
no tone busytone remote-onhook

Syntax Description

remote-onhook	Generates busy tone after remote onhook in basic call mode.

Command Default

Automatic busytone generation after remote disconnect is disabled.

Command Modes

Dial peer voice configuration (config-dial-peer)

Command History

Release	Modification
12.4(20)T	This command was introduced.

Usage Guidelines

The automatic busytone generation after remote disconnect in basic call mode feature is enabled and disabled per dial peer with the **tone busytone remote-onhook**command. The **tone busytone**command is available to all dial peer services. Each service determines whether to utilize or enable it. For STCAPP, only the Foreign eXchange Subscriber (FXS) loop-start port will enable this service.

Note The **tone busytone** command cannot coexist with thedialtone generation after remote-onhook feature. Because the **tone dialtone**is a default configuration, you must disable the feature using the **no tone dialtone**command before connfiguring the **tone busytone**command.

Use the **show dial-peer voice** command or the **show stcapp device voice** command to verify the feature is enabled.

Examples

The following example shows busytone generation after remote disconnect being configured:

```
Router(config-dial-peer)# tone busytone remote-onhook
```

Related Commands

Command	Description
show dial-peer voice	Displays information for voice dial peers.
tone dialtone	Enable automatic dial tone generation.
show stcapp device voice	Displays configuration information about STCAPP analog voice ports.

tone dialtone

To enable automatic dial-tone generation in basic call mode, use the **tone dialtone**command in dial peer configuration mode. To disable automatic dial-tone generation, use the **no** form of this command.

tone dialtone remote-onhook
no tone dialtone remote-onhook

Syntax Description

remote-onhook	Generates dial tone after remote onhook in basic call mode.

Command Default

Automatic dial-tone generation after remote disconnect is enabled.

Command Modes

Dial peer configuration (config-dial-peer)

Command History

Release	Modification
12.4(6)XE	This command was introduced.
12.4(11)T	This command was integrated into Cisco IOS Release 12.4(11)T.

Usage Guidelines

Use this command to generate immediate dial tone once a remote party disconnects, similar to what the user experiences in a PBX environment. If you disable this feature using the **no** form of this command, the user is required to go on hook or perform a hookflash to generate dial tone after the remote party disconnects in a basic two-part call scenario. This feature is supported on Skinny Client Control Protocol (SCCP) gateway controlled loop-start FXS ports only.

Examples

The following examples show that the automatic Dial Tone Generation After Remote Onhook feature is enabled. Because the dial tone generation after remote onhook feature is enabled by default, it does not display in the show running-config output.

```
Router# show running-config
service stcapp
 dial-peer voice 3001 pots
 port 1/1/1

Router# show dial-peer voice 3001
VoiceEncapPeer3001
 peer type = voice, system default peer = FALSE, information type = voice,
!
!
!
 in bound application associated: 'stcapp'
 dial tone generation after remote-onhook = enabled

Router# show stcapp device voice-port 1/1/1
Port Identifier:  1/1/1
!
Dialtone after remote-onhook feature: activated
```

The following examples show the dial tone generation after remote onhook feature disabled.

```
Router# show running-config
no tone dialtone remote-onhook
dial-peer voice 3002 pots
 service stcapp
 port 1/1/0
```

Related Commands

Command	Description
sccp	Enables SCCP and related applications.
show dial-peer voice	Displays information for voice dial peers.
show stcapp device	Displays configuration information about SCCP Telephony Control Application (STCAPP) analog voice ports.

tone incoming

To activate 2100-Hz answer (ANS) tone detection on either the IP or the PSTN side of the network and to disable the echo suppressor, use the **tone incoming**command in voice-service VoIP configuration mode or VoIP dial-peer configuration mode. To deactivate tone detection and disable the echo suppressor, use the **no** form of this command.

tone incoming [**ip** | **pstn**] {**ans-all auto-control** | **ans disable echo suppressor** | **anspr disable echo suppressor**}
no tone incoming

Syntax Description

ip	(Optional) Specifies tone detection on the IP side of the network.
pstn	(Optional) Specifies tone detection on the PSTN side of the network.
ans auto-control	Detects ANS tone and enables standard actions for modem tones.
ans-all disable echo suppressor	Detects modem answer tones and disables echo suppressor.
anspr disable echo suppressor	Detects /ANS tone and disables echo suppressor.

Command Default

Tone incoming detection is not enabled.

Command Modes

Voice-service VoIP configuration
VoIP dial-peer configuration

Command History

Release	Modification
12.3(14)T	This command was introduced.

Usage Guidelines

Use this command in voice-service VoIP or VoIP dial-peer configuration mode to activate detection of all ANS, ANSam, and ANSpr tones and enable or disable echo canceller control. When this command is issued in voice-service VoIP configuration mode*, all dial peers are globally configured unless a specific dial peer is configured for no tone incoming.*

To deactivate all 2100-Hz ANS, ANSam, and ANSpr tone detection on either the IP or the PSTN side of the network, and enable the echo canceller, use the **no tone incoming** command in voice-service VoIP configuration or VoIP dial-peer configuration mode.

If neither IP nor PSTN is specified, all ANS, ANSam, and ANSpr tones are detected on both sides of the network, and the echo suppressor is disabled in all cases.

The **tone incoming ip ans-all auto-control** command is equivalent to these two commands together:

- **tone incoming ip ans disable echo suppressor**
- **tone incoming ip anspr disable echo suppressor**

The **tone incoming pstn ans-all auto-control** command is equivalent to these two commands together:

- **tone incoming pstn ans disable echo suppressor**

- **tone incoming pstn anspr disable echo suppressor**

The **tone incoming ans-all auto-control** command is equivalent to these four commands together:

- **tone incoming ip ans disable echo suppressor**
- **tone incoming ip anspr disable echo suppressor**
- **tone incoming pstn ans disable echo suppressor**
- **tone incoming pstn anspr disable echo suppressor**

When modem tones from either the IP or PSTN direction are received, the echo canceller can be dynamically disabled to allow modem calls to pass through.

The IP tone detector feature applies only on the following NextPort platforms: Cisco AS5350, Cisco AS5400, and Cisco AS5850--and only with SIP and H.323 voice signaling. It does not apply to MGCP in VoIP dial-peer configuration mode.

The gateway must be configured for G.711 codecs for the IP tone detector feature to work (see the "Examples" section).

To display the status of the echo canceller, use the **show port operational status** command.

Examples

The following example configures tone detection of ANS tones in voice-service VoIP configuration mode:

```
Router(conf-voi-serv)# tone incoming ip ans disable echo supressor
```

The following example configures tone detection of all incoming ANS, ANSam, and ANSpr tones on a dial peer:

```
Router(config-dial-peer)# tone incoming ip ans-all auto-control
```

Related Commands

Command	Description
tone incoming system	Sets a dial peer for tone incoming or no tone incoming detection.
show port operational status	Displays the status of the echo canceller.

tone incoming system

To set a dial peer for tone incoming or no tone incoming, use the **tone incoming system** command in VoIP dial-peer configuration mode. To block the voice service VoIP settings for a dial peer, use the **no** form of this command.

tone incoming system
no tone incoming system

Syntax Description

This command has no arguments or keywords.

Command Default

The dial peer is set for tone incoming.

Command Modes

VoIP dial-peer configuration

Command History

Release	Modification
12.3(14)T	This command was introduced.

Usage Guidelines

Use this command in VoIP dial-peer configuration mode to activate or deactivate tone detection and to enable echo canceller control. When modem tones from either the IP or PSTN directions are received. The echo canceller can be dynamically disabled to allow modem calls through. This command is used primarily to allow or to block global voice service VoIP configuration settings.

To block the voice service VoIP settings for a dial peer, use the **no tone incoming system** command.

Examples

The following example shows activating tone detection for a dial peer.

```
Router(config-dial-peer)# tone incoming system
```

The following example shows deactivating tone detection for a dial peer.

```
Router(config-dial-peer)# no tone incoming system
```

Related Commands

Command	Description
tone incoming ans disable echo suppressor	Activates ANS tone detection.
tone incoming anspr disable echo canceller	Activates ANSpr tone detection.
tone incoming ans-all auto-control	Activates ANS, ANSam, and ANSpr tone detection.
show port operational status	Displays the status of the echo canceller.

tone ringback alert-no-PI

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

tone ringback alert-no-PI
no tone ringback alert-no-PI

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the Cisco 1700 series, Cisco 2600 series, Cisco 3600 series, Cisco 3700 series, Cisco 7200 series, Cisco AS5300, and Cisco AS5800.

Usage Guidelines

Use this command to generate ringback in an H.323 network when the attached device (for example, an ISDN device) cannot.

Examples

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

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

Related Commands

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

trace (voice service voip)

To configure the VoIP Trace framework in CUBE, use the **trace** command in voice service voip configuration mode. To disable VoIP tracing, use the **no** form of this command.

[no] trace

Command Default

Trace is enabled by default.

Command Modes

Voice Service VoIP configuration mode (conf-voi-serv)

Command History

Release	Modification
Cisco IOS XE Amsterdam 17.3.2 Cisco IOS XE Bengaluru 17.4.1a	This command was introduced on Cisco Unified Border Element.

Usage Guidelines

Use the **trace** command to configure the VoIP Trace framework to persistently monitor and troubleshoot SIP calls on CUBE. With **trace** enabled, event logging and debugging of VoIP parameters such as SIP messages, FSM, and Unified Communication flows processed by CUBE are logged.

VoIP tracing is disabled using the command **shutdown** under the **trace** configuration mode. To re-enable VoIP Trace, configure **[no] shutdown** . The **shutdown** command retains the custom **memory-limit** whereas **[no] trace** resets the **memory-limit** to default.

To define a custom limit for the memory allotted for storage of VoIP Trace information in CUBE, configure **memory-limit** *memory* under trace configuration mode. Range is 10–1000 MB. If **memory-limit** isn't configured, the default configuration of **memory-limit platform** is applied. By default, 10% of the total memory available to the IOS processor at the time of configuring the command will be reserved for VoIP Trace data storage.

Examples

The following is a sample configuration for enabling **trace** on Unified Border Element:

```
router#configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
router(config)#voice service voip
router(conf-voi-serv)#?
VOICE SERVICE configuration commands:
address-hiding Address hiding (SIP-SIP)
allow-connections Allow call connection types
call-quality Global call quality of service setup
callmonitor Call Monitoring
cause-code Sets the internal cause code for SIP and H323
clid Caller ID option
cpa Enable Call Progress Analysis for voip calls
default Set a command to its defaults
dtmf-interworking Dtmf Interworking
emergency List of Emergency Numbers
exit Exit from voice service configuration mode
fax Global fax commands
fax-relay Global fax relay commands
gcid Enable Global Call Identifcation for voip
h323 Global H.323 configuration commands
ip Voice service voip ip setup
lpcor Voice service voip lpcor setup
```

```
media Global media setting for voip calls
media-address Voice Media IP Address Range
mode Global mode setting for voip calls
modem Global modem commands
no Negate a command or set its defaults
notify send facility indication to application
qsig QSIG
redirect voip call redirect
redundancy-group Associate redundancy-group with voice HA
redundancy-reload Reload control when RG fail
rtcp Configure RTCP report generation
rtp-media-loop Global setting for rtp media loop count
rtp-port Global setting for rtp port range
shutdown Stop VoIP services gracefully without dropping active calls
signaling Global setting for signaling payload handling
sip SIP configuration commands
srtp Allow Secure calls
stun STUN configuration commands
supplementary-service Config supplementary service features
trace Voip Trace configuration
voice enable voice parameters
vpn-group Enter vpn-group mode
vpn-profile Enter vpn-profile mode

router(conf-voi-serv)# trace
```

Related Commands

Command	Description
memory-limit (trace)	Defines the memory limit for storing VoIP Trace information.
shutdown (trace)	Disable the VoIP Trace serviceability framework in CUBE.
show voip trace	Displays the VoIP Trace information for SIP legs on a call received on CUBE

transfer

To define a Feature Access Code (FAC) to access the Call Transfer feature in feature mode on analog phones connected to FXS ports, use the **transfer** command in STC application feature-mode call-control configuration mode. To return the code to its default, use the **no** form of this command.

transfer *keypad-character*
no transfer

Syntax Description

keypad-character	Character string of one to four characters that can be dialed on a telephone keypad (0-9, *, #). Default is #2.

Command Default

The default value is #2.

Command Modes

STC application feature-mode call-control configuration (config-stcapp-fmcode)

Command History

Release	Modification
15.0(1)M	This command was introduced.

Usage Guidelines

This command changes the value of the FAC for Call Transfer from the default (#2) to the specified value.

If you attempt to configure this command with a value that is already configured for another FAC in feature mode, you receive a message. This message will not prevent you from configuring the feature code. If you configure a duplicate FAC, the system implements the first feature it matches in the order of precedence as determined by the value for each FAC (#1 to #5).

If you attempt to configure this command with a value that precludes or is precluded by another FAC in feature mode, you receive a message. If you configure a FAC to a value that precludes or is precluded by another FAC in feature mode, the system always executes the call feature with the shortest code and ignores the longer code. For example, 1 will always preclude 12 and 123. These messages will not prevent you from configuring the feature code. You must configure a new value for the precluded code in order to enable phone user access to that feature.

Examples

The following example shows how to change the value of the feature code for the Call Transfer feature from the default (#2). With this configuration, a phone user presses hook flash to get the first dial tone, then dials an extension number to connect to a second call. When the second call is established, the user presses hook flash to get a feature tone and then dials 22 to transfer the call; the user hears silence after the call is transferred.

```
Router(config)# stcapp call-control mode feature
Router(config-stcapp-fmcode)# transfer 22
Router(config-stcapp-fmcode)# exit
```

Related Commands

Command	Description
conference	Defines FAC in Feature Mode to initiate a three-party conference.

Command	Description
drop-last-conferee	Defines FAC in feature mode to use to drop last active call during a three-party conference.
hangup-last-active-call	Defines FAC in feature mode to drop last active call during a three-party conferencee.
toggle-between-two-calls	Defines FAC in feature mode to toggle between two active calls.

translate

To apply a translation rule to manipulate dialed digits on an inbound POTS call leg, use the **translate** command in voice-port configuration mode. To remove the translation rule, use the **no** form of this command.

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

Syntax Description

calling -number	Translation rule applies to the inbound calling party number.
called -number	Translation rule applies to the inbound called party number.
name -tag	Tag number by which the rule set is referenced. This is an arbitrarily chosen number. Range is from 1 to 2147483647. There is no default value.

Command Default

No default behavior or values

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for VoIP on Cisco AS5300.
12.0(7)XK	This command was implemented for VoIP on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T and implemented for VoIP Cisco AS5300, Cisco 7200, and Cisco 7500.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.

Usage Guidelines

A translation rule is a general-purpose digit-manipulation mechanism that performs operations such as automatically adding telephone area and prefix codes to dialed numbers.

Examples

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

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

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

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

Related Commands

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

translate (translation profiles)

To associate a translation rule with a voice translation profile, use the **translate** command in voice translation-profile configuration mode. To delete the translation rule from the profile, use the **no** form of this command.

translate {**called** | **calling** | **redirect-called** | **redirect-target**} *translation-rule-number*
no translate {**called** | **calling** | **redirect-called** | **redirect-target**} *translation-rule-number*

Syntax Description

called	Associates the translation rule with called numbers.
calling	Associates the translation rule with calling numbers.
redirect -called	Associates the translation rule with redirected called numbers.
redirect-target	Associates the translation rule with transfer-to numbers and call-forwarding final destination numbers.
translation -rule-number	Number of the translation rule to use for the call translation. Valid range is from 1 to 2147483647. There is no default value.

Command Default

No translation rule is associated with the translation profile.

Command Modes

Voice translation-profile configuration (cfg-translation-profile)

Command History

Release	Modification
12.0(7)XR1	This command was introduced on the Cisco AS5300.
12.0(7)XK	This command was implemented on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T and implemented on the following platforms: Cisco 1750, Cisco AS5300, Cisco 7200 series, and Cisco 7500 series.
12.1(2)T	This command was implemented on the Cisco MC3810.
12.2(11)T	This command was reconfigured for voice translation-profile configuration mode. The **redirect-called**keyword and *translation-rule-number*argument were added.
12.4(11)XJ	The **redirect-target** keyword was added.
12.4(15)T	The **redirect-target** keyword was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Use this command as part of a voice translation-profile definition. Enter this command for each translation rule that is part of the profile definition.

Examples

The following example defines voice translation profile "sjmorning" with two translation rules: translation rule 15 for called numbers and translation rule 36 for calling numbers.

```
Router(config)# voice translation-profile sjmorning
Router(cfg-translation-profile)# translate called 15
Router(cfg-translation-profile)# translate calling 36
```

Related Commands

Command	Description
rule (voice translation-rule)	Sets the criteria for the translation-rule.
show voice translation-profile	Displays the configuration of the translation-profile.
translation-profile (dial-peer)	Assigns a translation profile to a dial peer.
translation-profile (source group)	Assigns a translation profile to a source IP group.
translation-profile (trunk group)	Assigns a translation profile to a trunk group.
translation-profile (voice port)	Assigns a translation profile to a voice port.
translation-profile (voice service POTS)	Assigns a translation profile to an NFAS interface.
voice translation-profile	Initiates the translation-profile definition.
voice translation-rule	Initiates the translation-rule definition.

translate-outgoing

To apply a translation rule to manipulate dialed digits on an outbound POTS or VoIP call leg, use the **translate-outgoing command in**dial-peer configuration mode. To disable the translation rule, use the **no** form of this command.

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

Syntax Description

calling -number	Apply to the outbound calling party number.
called -number	Apply to the outbound called party number.
name -tag	Tag number by which the rule set is referenced. This is an arbitrarily chosen number. Range is 1 to 2147483647. There is no default value.

Command Default

No default behavior or values

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for VoIP on Cisco AS5300.
12.0(7)XK	This command was implemented for VoIP on the Cisco 2600 series, Cisco 3600 series, and Cisco MC3810.
12.1(1)T	This command was integrated into Cisco IOS Release 12.2(1)T and implemented for VoIP on the Cisco 1750, Cisco AS5300, Cisco 7200, and Cisco 7500. support for the Cisco MC3810 is not included in this release.
12.1(2)T	This command is supported on the Cisco MC3810 in this release.

Examples

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

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

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

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

Related Commands

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

translation-profile (dial peer)

To assign a translation profile to a dial peer, use the **translation-profile** command in dial peer configuration mode. To delete the translation profile from the dial peer, use the **no** form of this command.

translation-profile {**incoming** | **outgoing**} *name*
no translation-profile {**incoming** | **outgoing**} *name*

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
outgoing	Specifies that this translation profile handles outgoing calls.
name	Name of the translation profile.

Command Default

No default behavior or values

Command Modes

Dial peer configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.
12.4(22)T	Support for IPv6 was added.
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

Usage Guidelines

Use the **translation-profile** command to assign a predefined translation profile to a dial peer.

Examples

The following example assigns the translation profile named "profile1" to handle translation of outgoing calls for a dial peer:

```
Router(config)# dial-peer voice 111 pots
Router(config-dial-peer)# translation-profile outgoing profile1
```

Related Commands

Command	Description
rule (voice translation-rule)	Sets the criteria for the translation rule.
show voice translation-profile	Displays the configuration of a translation profile.
translate (translation profiles)	Assigns a translation rule to a translation profile.
voice translation-profile	Initiates the translation-profile definition.
voice translation-rule	Initiates the translation-rule definition.

translation-profile (source group)

To assign a translation profile to a source IP group, use the **translation-profile** command in source group configuration mode. To delete the translation profile from the source IP group, use the **no** form of this command.

translation-profile incoming *name*
no translation-profile incoming *name*

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
name	Name of the translation profile.

Command Default

No default behavior or values

Command Modes

Source group configuration

Command History

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

Usage Guidelines

Use the **translation-profile** command to assign a predefined translation profile to a source IP group.

Examples

The following example assigns the translation profile named "chicago" to handle translation of incoming calls for a voice source group:

```
Router(config)# voice source-group alpha
Router(cfg-source-grp)# translation-profile incoming chicago
```

Related Commands

Command	Description
rule (voice translation-rule)	Sets the criteria for the translation rule.
show voice translation-profile	Displays the configuration of a translation profile.
translate (translation profiles)	Assigns a translation rule to a translation profile.
voice translation-profile	Initiates the translation-profile definition.
voice translation-rule	Initiates the translation-rule definition.

translation-profile (trunk group)

To assign a translation profile to a trunk group, use the **translation-profile** command in trunk group configuration mode. To delete the translation profile from the trunk group, use the **no** form of this command.

translation-profile {**incoming** | **outgoing**} *name*
no translation-profile {**incoming** | **outgoing**} *name*

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
outgoing	Specifies that this translation profile handles outgoing calls.
name	Name of the translation profile.

Command Default

No default behavior or values

Command Modes

Trunk group configuration

Command History

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

Usage Guidelines

Use the **translation-profile** command to assign a predefined translation profile to a trunk group.

Examples

The following example assigns the translation profile named "newyork" to handle translation of incoming calls for a trunk group:

```
Router(config)# trunk group 10
Router(config-trunk-group)# translation-profile incoming newyork
```

Related Commands

Command	Description
rule (voice translation-rule)	Sets the criteria for the translation rule.
show voice translation-profile	Displays the configuration of a translation profile.
translate (translation profiles)	Assigns a translation rule to a translation profile.
voice translation-profile	Initiates the translation-profile definition.
voice translation-rule	Initiates the translation-rule definition.

translation-profile (voice port)

To assign a translation profile to a voice port, use the **translation-profile** command in voice port configuration mode. To delete the translation profile from the voice port, use the **no** form of this command.

translation-profile {**incoming** | **outgoing**} *name*
no translation-profile {**incoming** | **outgoing**} *name*

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
outgoing	Specifies that this translation profile handles outgoing calls.
name	Name of the translation profile.

Command Default

No default behavior or values

Command Modes

Voice port configuration

Command History

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

Usage Guidelines

Use the **translation-profile** command to assign a predefined translation profile to a voice port.

Examples

The following example assigns the translation profile named "chicago" to handle translation of incoming calls and a translation profile named "sanjose" to handle outgoing calls for a voice port:

```
Router(config)# voice-port 1/0/0
Router(config-voiceport)# translation-profile incoming chicago
Router(config-voiceport)# translation-profile outgoing sanjose
```

Related Commands

Command	Description
rule (voice translation-rule)	Sets the criteria for the translation rule.
show voice translation-profile	Displays the configuration of a translation profile.
translate (translation profiles)	Assigns a translation rule to a translation profile.
voice translation-profile	Initiates the translation-profile definition.
voice translation-rule	Initiates the translation-rule definition.

translation-profile (voice service POTS)

To assign a translation profile to a non-facility associated signaling (NFAS) interface, use the **translation-profile** command in voice service POTS configuration mode. To delete the translation profile from the interface, use the **no** form of this command.

translation-profile [**incoming** | **outgoing**] **controller** [**T1** | **E1**] *unit-number name*
no translation-profile [**incoming** | **outgoing**] **controller** [**T1** | **E1**] *unit-number name*

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
outgoing	Specifies that this translation profile handles outgoing calls.
T1	T1 controller.
E1	E1 controller.
unit-number	Number of the controller unit.
name	Name of the translation profile.

Command Default

No default behavior or values

Command Modes

Voice service POTS configuration

Command History

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

Usage Guidelines

Use the **translation-profile** command to assign a predefined translation profile to an NFAS interface.

Examples

The following example assigns to an NFAS interface the translation profile named "delta1" to outgoing T1 calls on controller slot 3 and translation profile "alpha" to incoming T1 calls on controller slot 2:

```
Router(config)# voice service pots
Router(conf-voi-serv)# translation-profile outgoing controller T1 3 delta1
Router(conf-voi-serv)# translation-profile incoming controller T1 2 alpha
```

Related Commands

Command	Description
rule (voice translation-rule)	Sets the criteria for the translation rule.
show voice translation-profile	Displays the configuration of a translation profile.
translate (translation profiles)	Assigns a translation rule to a translation profile.
voice translation-profile	Initiates the translation-profile definition.

Command	Description
voice translation-rule	Initiates the translation-rule definition.

translation-rule

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

translation-rule *name-tag*
no translation-rule *name-tag*

Syntax Description

name -tag	Tag number by which the rule set is referenced. This is an arbitrarily chosen number. Range is from 1 to 2147483647. There is no default value.

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced for VoIP on Cisco AS5300.
12.0(7)XK	This command was implemented for the following voice technologies on the following platforms: • VoIP Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 • VoFR Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 • VoATM Cisco 3600 series and Cisco MC3810
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T and implemented for the following voice technology on the following platforms: VoIP (Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco 7200 series, and Cisco 7500 series)
12.1(2)T	This command was integrated ino Cisco IOS Release 12.1(2)T for the following voice technologies on the following platforms: • VoIP Cisco MC3810 • VoFR Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 • VoATM Cisco 3600 series and Cisco MC3810
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

This command applies to all translation rules.

Examples

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

```
translation-rule 21

   rule 1 555.% 1408555 subscriber international
```

Related Commands

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

transport (sip-ua)

To configure the Session Initiation Protocol (SIP) user agent (gateway) for SIP signaling messages on inbound calls through the SIP TCP, Transport Layer Security (TLS) over TCP, or User Datagram Protocol (UDP) socket, use the **transport** command in SIP user agent configuration mode. To block reception of SIP signaling messages on a particular socket, use the **no** form of this command.

transport { **tcp** [**tls** [**v1.0** | **v1.1** | **v1.2** [**minimum**] | **v1.3**]] | **udp** }
no transport { **tcp** [**tls** [**v1.0** | **v1.1** | **v1.2** [**minimum**] | **v1.3**]] | **udp** }
default transport { **tcp** [**tls**] | **udp** }

Syntax Description

tcp	SIP user agent receives SIP messages on TCP port 5060.
tls	(Optional) SIP user agent receives SIP messages on TLS over TCP port 5061. You can configure TLS version 1.0, 1.1, 1.2, or 1.3. Starting from Cisco IOS XE 26.1.1 release, insecure TLS versions and ciphers are not supported in default configurations. However, these insecure configurations are supported in "system mode insecure" operation-mode.
minimum	(Optional) Specifies minimum configured TLS version. The **minimum** keyword can be configured only with TLS version 1.2. This configuration enables TLS versions 1.2 and 1.3.
udp	SIP user agent receives SIP messages on UDP port 5060.

Command Default

TCP, TLS over TCP, and UDP transport protocols are enabled.

Command Modes

SIP user-agent configuration (config-sip-ua)

Command History

Release	Modification
Cisco IOS XE 26.1.1	This command is modified with the removal of insecure TLS versions (1.0, 1.1) and associated ciphers from default configurations. Support for insecure configurations in "system mode insecure" operation-mode.
Cisco IOS XE 17.18.2	Security warnings for usage of legacy TLS and associated weaker ciphers.
Cisco IOS XE 17.14.1a	This command is modified to include the TLS version 1.3 support. In addition, **minimum** keyword configuration support is introduced with TLS version 1.2. Introduced support for the following YANG models: • **transport tcp tls v1.3** • **transport tcp tls v1.2 minimum**
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
15.6(1)T and 3.17S	This command was modified to include the TLS version 1.2.

Release	Modification
12.4(6)T	The optional TLS keyword was added to the command.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms in this release.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 7200 series routers. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms were not included in this release.
12.2(2)XB1	This command was implemented on Cisco AS5850 platforms.
12.2(2)XA	This command was implemented on Cisco AS5400 and Cisco AS5350 platforms.
12.1(3)T	This command was integrated into Cisco IOS Release 12.1(3)T.
12.1(1)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco AS5300 platforms.

Usage Guidelines

This command controls whether messages reach the SIP service provider interface (SPI). Setting **tcp**, or **tls** over **tcp**, or **udp** as the protocol for the SIP user agents to listen on port 5060.

To block reception of SIP signaling messages on a specific socket, use the **no** form of this command.

To reset this command to the default value, use the **default** form of this command.

Note In the Cisco IOS XE 17.18.2 release, a security warning message appears for configurations using TLS versions below 1.2 and associated weaker ciphers. For secure configurations, we recommend configuring stronger ciphers with TLS version 1.2 or higher.

Note Starting from Cisco IOS XE 26.1.1 release, the insecure TLS versions (1.0, 1.1) and associated ciphers are not supported in default configurations. However, insecure configurations are allowed in "system mode insecure" operation-mode. The following are the insecure ciphers associated with TLS versions (1.0, 1.1):

- DHE_RSA_WITH_AES_128_CBC_SHA
- RSA_WITH_AES_128_CBC_SHA
- RSA_WITH_AES_256_CBC_SHA

Examples

The following example sets the SIP user agent to allow the reception of SIP signaling messages on the UDP socket:

```
sip-ua
 transport udp
```

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

```
sip-ua
 transport tcp
```

The following example sets the SIP user agent to allow the reception of SIP signaling messages on the TLS over TCP socket:

```
sip-ua
 transport tcp tls
  v1.0  Enable TLS Version 1.0
  v1.1  Enable TLS Version 1.1
  v1.2  Enable TLS Version 1.2
  v1.3  Enable TLS Version 1.3
```

The following example sets the SIP user agent to TLS version 1.2 as minimum, enabling both TLS versions 1.2 and 1.3:

```
sip-ua
 transport tcp tls v1.2 minimum
```

The following example illustrates a security warning message display in Cisco IOS XE 17.18.2 release for configurations using TLS versions below 1.2:

```
Device(config-sip-ua)#transport tcp tls v1.0
 SECURITY WARNING - Module: SIPUA, Command: transport tcp tls v1.0, Reason: Weak tls version,

Remediation: Use stronger tls version to enhance security

Device(config-sip-ua)#do sh run | sec sip-ua
sip-ua
 no remote-party-id
 timers connect 501
 timers dns registrar-cache 60
 transport tcp tls v1.0
 registration spike 100
 connection-reuse
  crypto signaling remote-addr 10.10.10.70 255.255.255.255  trustpoint CUBE-TLS strict-cipher

alias exec tcp show sip-ua connections tcp tls detail
```

The following example illustrates weaker TLS versions (1.0, 1.1) configuration in insecure mode with warning message display, starting from Cisco IOS XE 26.1.1 release:

```
Device# configure terminal
Device(config)# system mode insecure
Device(config)# sip-ua
Device(config-sip-ua)# transport tcp tls v1.1
SECURITY WARNING - Module: SIPUA, Command: transport tcp tls v1.0, Reason: Weak tls version,

Remediation: Use stronger tls version to enhance security
```

The following example illustrates weaker TLS versions (1.0, 1.1) configuration in secure mode with error message display, starting from Cisco IOS XE 26.1.1 release:

```
Device(config-sip-ua)# transport tcp tls v1.1
%Error:Insecure configurations are not permitted in secure mode. To proceed, set the system
 mode to insecure
using the command system mode insecure, and then try again.
Module: SIPUA, Command: transport tcp tls v1.1 , Reason: Weak tls version, Remediation: Use
 stronger tls version
to enhance security
%ERROR: Security policy check failed, configuration can't be applied
```

Related Commands

Command	Description
sip-ua	Enables the SIP user agent configuration commands.

transport switch

To enable switching between UDP and TCP transport mechanisms globally for large Session Initiation Protocol (SIP) messages, use the **transport switch**command in SIP configuration mode. To disable switching between UDP and TCP transport mechanisms globally for large SIP messages, use the **no** form of this command.

transport switch udp tcp
no transport switch udp tcp

Syntax Description

udp	Enables switching the transport mechanism from UDP on the basis of the size of the SIP request being greater than the MTU size.
tcp	Enables switching transport to TCP.

Command Default

Disabled.

Command Modes

SIP configuration

Command History

Release	Modification
12.3(8)T	This command was introduced.

Usage Guidelines

Switching between transports is provided globally on the router and also on an individual VoIP dial peer.

- Dial-peer mode. You can configure transport for a specific dial peer by using the **voice-class sip transport switch**command. The **voice-class sip transport switch command in** dial-peer configuration mode takes precedence over **the transport switch**command in **global configuration mode.**
- SIP mode. You can configure transport globally by using the **transport switch**command. The **transport switch** command is considered only when there is no matching VoIP dial peer.

In a call forking scenario, if this command is configured, the configuration applies to all forks.

Examples

The following example enables switching of the transport from UDP to TCP:

```
Router(config)# voice service voip
Router(config-voi-srv)# sip
Router(conf-serv-sip)# transport switch udp tcp
```

Related Commands

Command	Description
debug ccsip transport	Enables tracing of the SIP transport handler and the TCP or UDP process.
sip	Enters SIP configuration mode from voice-service VoIP configuration mode.
voice -class sip transport switch	Enables switching between transport mechanisms if the SIP message is larger than 1300 bytes for a specific dial peer.

trunk group (global)

To define or modify the definition of a trunk group and to enter trunk group configuration mode, use the **trunk group**command in global configuration mode. To delete the trunk group, use the **no** form of this command.

trunk group *name*
no trunk group *name*

Syntax Description

name	Name of the trunk group. Valid names contain a maximum of 63 alphanumeric characters.

Command Default

No trunk group is defined.

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the **trunk group** command to assign a number or a name to a set of trunk characteristics. The set of characteristics, or *profile* , is assigned to specific trunks as part of the usual trunk configuration steps.

The **trunk group** command initiates the profile definition and switches from global configuration to trunk group configuration mode. Additional commands are available to construct the characteristics of the profile.

Up to 1000 trunk groups can be configured on the gateway provided that the gateway has sufficient memory to store the profiles. If you see the message "Trunk group name could not be added as the threshold has been reached", enter the **debug tgrm**command and check the number of trunk groups or check for insufficient memory.

To associate a trunk group with an interface, use the **trunk-group** (interface) command. A trunk group that was created using the **trunk group** (global) command can be associated with an interface. However, a trunk group need not be defined globally before being associated with an interface. If a trunk group has not been defined globally, it will be created by issuing the **trunk-group** (interface) command.

Examples

The following example creates trunk group 5 and configures the trunk group profile:

```
Router(config)# trunk group 5
Router(config-trunk-group)# carrier-id allcalls
Router(config-trunk-group)# max-calls voice 500 in
Router(config-trunk-group)# hunt-scheme round-robin even up
Router(config-trunk-group)# translation-profile incoming 3
Router(config-trunk-group)# translation-profile outgoing 2
Router(config-trunk-group)# exit
```

The following example creates a trunk group named "mytrunk" and configures the trunk group profile:

```
Router(config)# trunk group mytrunk
Router(config-trunk-group)# carrier-id local
Router(config-trunk-group)# max-calls voice 500
```

```
Router(config-trunk-group)# hunt-scheme least-idle
Router(config-trunk-group)# translation-profile incoming 1
Router(config-trunk-group)# translation-profile outgoing 12
Router(config-trunk-group)# exit
```

Related Commands

Command	Description
carrier-id (trunk group)	Identifies the carrier that owns the trunk group.
description (trunk group)	Permits a description to be associated with a trunk group.
hunt-scheme least-idle	Specifies the least-idle channel search method for incoming and outgoing calls.
hunt-scheme least-used	Specifies the least-used channel search method for incoming and outgoing calls.
hunt-scheme longest-idle	Specifies the longest-idle channel search method for incoming and outgoing calls.
hunt-scheme random	Specifies the random channel search method for incoming and outgoing calls.
hunt-scheme round-robin	Specifies the round-robin channel search method for incoming and outgoing calls.
hunt-scheme sequential	Specifies the sequential channel search method for incoming and outgoing calls.
max-calls	Specifies the number of incoming and outgoing voice and data calls that a trunk group can handle.
show trunk group	Displays the configuration of trunk groups.
translation-profile (trunk group)	Defines call number translation profiles for incoming and outgoing calls.
trunk-group (interface)	Assigns an ISDN PRI or NFAS interface to a trunk group.

trunk-group (CAS custom)

To assign a channel-associated signaling (CAS) trunk to a trunk group, use the **trunk-group**command in CAS custom configuration mode. To delete the CAS trunk from the trunk group, use the **no** form of this command.

trunk-group *name* [*preference-num*]
no trunk-group *name* [*preference-num*]

Syntax Description

name	Name of the trunk group. Maximum length of the trunk group name is 63 alphanumeric characters.
preference -num	(Optional) Priority of the trunk group member in a trunk group. Range is from 1 (highest priority) to 64 (lowest priority).

Command Default

Preference-num is set lower than 64 (internally set to 65)

Command Modes

CAS custom configuration

Command History

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

Usage Guidelines

Use the **trunk-group** command to assign a CAS trunk as a member of a trunk group. This assignment provides the CAS trunk with carrier information, a hunt scheme for finding an available channel for the outgoing call, and translation profiles for number translation.

If more than one CAS trunk is assigned to the same trunk group, the *preference-num* value determines the order in which the trunk group uses the interfaces. A *preference-num* value of 1 is the highest preference so that the trunk is used first; a value of 64 is the lowest preference so that the trunk is used last. If no value is entered for *preference-num*, the software assigns the trunk a preference of 65, which causes that trunk to be used after all other trunks are used.

If two CAS trunks have the same *preference-num*, the trunk that was configured first is used before the other trunk.

A CAS trunk can belong to only one trunk group.

If an interface is removed from the CAS trunk, the interface is removed automatically from the trunk group. A new nonprimary CAS interface is automatically a member of the same trunk group as its primary CAS interface.

Examples

The following example assigns two CAS interfaces to trunk group "westcoast". The preference value for DS0 group 2 is lower than for DS0 group 1; hence DS0 group 2 has a higher priority. Trunk group "westcoast" uses DS0 group 2 first.

```
Router(config)# controller T1 1/0
Router(config-controller)# ds0-group 1 timeslots 1-10 type e&m-fgd
Router(config-controller)# cas-custom 1
```

```
Router(config-controller)# trunk-group westcoast 5
Router(config-controller)# exit
Router(config)# controller T1 1/0
Router(config-controller)# ds0-group 2 timeslots 15-20 type e&m-fgd
Router(config-controller)# cas-custom 2
Router(config-controller)# trunk-group westcoast 3
Router(config-controller)# exit
```

Related Commands

Command	Description
show trunk group	Displays the configuration of a trunk group.

trunkgroup (dial peer)

To assign a dial peer to a trunk group for trunk group label routing, use the **trunkgroup**command in dial-peer configuration mode. To delete the dial peer from the trunk group, use the **no** form of this command.

trunkgroup *name preference-num*
no trunkgroup *name*

Syntax Description

name	Label of the trunk group to use for the call. Valid trunk group names contain a maximum of 63 alphanumeric characters.
preference -num	Preference or priority of the trunk group. Range is from 1 (highest priority) to 64 (lowest priority).

Command Default

Preference-num is set lower than 64 (internally set to 65)

Command Modes

Dial peer configuration (config dial-peer)

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2	This command was integrated into the Cisco IOS Release 12.2.
12.2(11)T	The*preference -num* argument was added.

Usage Guidelines

Use the **trunkgroup** command to assign an outgoing dial peer as a member of one or more trunk groups. This assignment provides the dial peer with carrier information, a hunt scheme for finding an available channel for the outgoing call, and translation profiles for number translation.

If the dial peer is a member of more than one trunk group, use the *preference-num* value to set the order in which the trunk groups will be used for the dial peer. A *preference-num* value of 1 is the highest preference so that the trunk group is used first; a value of 64 is the lowest preference so that the trunk group is used last. If no value is entered for *preference-num*, the software assigns the trunk group a preference of 65, which causes that trunk group to be selected after all other trunks are used.

If two trunk groups have the same *preference-num*, the trunk group that was configured first is used before the other trunk group.

Examples

In the following example, dial peer 112 should use the trunk group "east17" and trunk group "north5" for outbound dial peer matching. When selecting a trunk group, "north5" is used first because it has a higher preference than "east17":

```
Router(config)# dial-peer voice 112 pots
Router(config-dial-peer)# trunkgroup east17 3
Router(config-dial-peer)# trunkgroup north5 1
```

Related Commands

Command	Description
debug dialpeer	Initiates dial peer debugging.
show dial-peer voice	Displays the dial peer configuration.
translation-profile (dial peer)	Defines call number translation profiles for incoming and outgoing calls.

trunk-group (interface)

To assign an ISDN PRI or Non-Facility Associated Signaling (NFAS) interface to a trunk group, use the **trunk-group**command in interface configuration mode. To delete the interface from the trunk group, use the **no** form of this command.

trunk-group *name* [*preference-num*]
no trunk-group *name* [*preference-num*]

Syntax Description

name	Name of the trunk group. Valid trunk group names contain a maximum of 63 alphanumeric characters.
preference -num	Priority of the trunk group member in a trunk group. Range is from 1 (highest priority) to 64 (lowest priority).

Command Default

Preference-num is set lower than 64 (internally set to 65)

Command Modes

Interface configuration

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2	This command was integrated into Cisco IOS Release 12.2.
12.2(11)T	The trunk-group identification was expanded to include alphanumeric characters using the *name* **argument, and the** *preference-num* argument was added.

Usage Guidelines

Use the **trunk-group** command to configure an ISDN PRI or Non-Facility Associated Signaling (NFAS) interface as a member of a trunk group. This assignment provides the interface with carrier information, a hunt scheme for finding an available channel for the outgoing call, and translation profiles for number translation.

If more than one interface is assigned to the same trunk group, the *preference_num* value determines the order in which the trunk group uses the interfaces. A *preference-num* value of 1 is the highest preference so that the interface is used first; a value of 64 is the lowest preference so that the interface is used last. If no value is entered for *preference-num*, the software assigns the interface a preference of 65, which causes that interface to be selected after all other interfaces are used.

If two interfaces have the same *preference-num*, the interface that was configured first is used before the other interface.

An interface can belong to only one trunk group. Multiple interfaces can belong to the same trunk group.

If an NFAS interface group is assigned as a member of a trunk group, all the subinterfaces belong to that trunk group.

If a subinterface is removed from the NFAS group, the subinterface is removed automatically from the trunk group.

If a new nonprimary NFAS interface is added to the NFAS group, that interface automatically becomes a member of the same trunk group as its primary NFAS interface.

Examples

The following example assigns an ISDN interface to trunk group "eastern" with a preference of 3.

```
Router(config)# interface Serial2:23
Router(config-if)# no ip address
Router(config-if)# isdn switch-type primary-ni
Router(config-if)# isdn T306 30000
Router(config-if)# isdn T310 10000
Router(config-if)# no cdp enable
Router(config-if)# trunk-group eastern 3
Router(config-if)# exit
```

If another interface were assigned to trunk group "eastern" with preference of 1 or 2, the trunk group would use that interface before the one shown above.

Related Commands

Command	Description
show trunk group	Displays the configuration of the trunk group.

trunk-group (voice port)

To assign an analog voice port to a trunk group, use the **trunk-group**command in voice port configuration mode. To delete the trunk group, use the **no** form of this command.

trunk-group *name* [*preference-num*]
no trunk-group *name* [*preference-num*]

Syntax Description

name	Name of the trunk group. Maximum length of the trunk group name is 63 alphanumeric characters.
preference -num	Priority of the trunk group member in a trunk group. Range is from 1 (highest priority) to 64 (lowest priority).

Command Default

Preference-num is set lower than 64 (internally set to 65)

Command Modes

Voice port configuration

Command History

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

Usage Guidelines

Use the **trunk-group** command to configure an analog voice port as a member of a trunk group. This assignment provides the voice port with carrier information, a hunt scheme for finding an available channel for the outgoing call, and translation profiles for number translation.

If more than one voice port is assigned to the same trunk group, the *preference-num* value determines the order by which the trunk group uses the voice ports. A *preference-num* value of 1 is the highest preference so that the voice port is used first; a value of 64 is the lowest preference so that the voice port is used last. If no value is entered for *preference-num*, the software assigns the voice port a preference of 65, which causes that voice port to be selected after all other voice ports are used.

If two voice ports have the same *preference-num*, the voice port that was configured first is used before the other voice port.

A voice port can belong to only one trunk group. Multiple voice ports can belong to the same trunk group.

Examples

The following example assigns voice port 1/0/0 and voice port 1/0/1 to trunk group "north5". Trunk group "north5" uses voice port 1/0/1 before using voice port 1/0/0 because voice port 1/0/1 has preference 1, which is a higher priority than voice port 1/0/0, with preference 2.

```
Router(config)# voice port 1/0/0
Router(config-voiceport)# translation-profile incoming 7
Router(config-voiceport)# translation-profile outgoing 4
Router(config-voiceport)# trunk-group north5 2
Router(config-voiceport)# exit
Router(config)# voice port 1/0/1
Router(config-voiceport)# translation-profile incoming 3
Router(config-voiceport)# translation-profile outgoing 8
```

```
Router(config-voiceport)# trunk-group north5 1
Router(config-voiceport)# exit
```

Related Commands

Command	Description
show trunk group	Displays the configuration of a trunk group.

trunk-group-label (dial peer)

To specify a trunk group as the source or target of a call, use the **trunk-group-label**command in dial peer configuration mode. To delete the trunk group label, use the **no** form of this command.

trunk-group-label {**source** | **target**} *name*
no trunk-group-label {**source** | **target**} *name*

Syntax Description

source	Indicates the trunk group as the source of the incoming call.
target	Indicates the trunk group as the target of the outbound call.
name	Trunk group label. Maximum length of the trunk group label is 127 alphanumeric characters.

Command Default

No default behavior or values

Command Modes

Dial peer configuration (config dial-peer)

Command History

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

Usage Guidelines

An originating gateway uses the source trunk group label as a matching key to route the call over an inbound dial peer. The terminating gateway uses the target trunk group label to select a dial peer for routing the outbound call over a POTS line.

If a dial peer has a source (or target) carrier ID already defined, then assigning a source (or target) trunk group label to that same dial peer overrides the source (or target) carrier ID. The same is true for the reverse: if a dial peer has a source (or target) trunk group label defined, then assigning a source (or target) carrier ID for that same dial peer overrides the source (or target) trunk group label.

The name of a trunk group label and carrier ID cannot be the same in dial peers.

Examples

The following example shows that dial peer 112 should use trunk group label "north3" for inbound dial peer matching and trunk group label "east17" for outbound dial peer matching:

```
Router(config)# dial-peer voice 112 pots
Router(config-dial-peer)# trunk-group-label source north3
Router(config-dial-peer)# trunk-group-label target east17
```

Related Commands

Command	Description
carrier-id (dial peer)	Specifies the carrier associated with a VoIP call.
show dial-peer voice	Displays configuration information for dial peers.

trunk-group-label (voice source group)

To define a trunk group label in a source IP group, use the **trunk-group-label**command in voice source group configuration mode. To delete the trunk group label, use the **no** form of this command.

trunk-group-label {**source** | **target**} *name*
no trunk-group-label {**source** | **target**} *name*

Syntax Description

source	Indicates the trunk group as the source of the incoming call.
target	Indicates the trunk group as the target of the outbound call.
name	Trunk group label. Maximum length of the trunk group label is 127 alphanumeric characters.

Command Default

No default behavior or values

Command Modes

Voice source group configuration

Command History

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

Usage Guidelines

A terminating gateway uses the source trunk group label as a search key to find a source IP group for the incoming VoIP call. The gateway uses the target trunk group label to select an outbound dial peer to route the call over a POTS line.

If a source IP group has a source (or target) carrier ID already defined, then assigning a source (or target) trunk group label to that same source IP group overrides the source (or target) carrier ID. The same is true for the reverse: if a source IP group has a source (or target) trunk group label defined, then assigning a source (or target) carrier ID for that same source IP group overrides the source (or target) trunk group label.

The name of a trunk group label and carrier ID of the same type (source or target) cannot be the same in the source IP group.

Examples

The following example shows that source IP group "alpha" uses trunk group "north3" to search for a source IP group for incoming VoIP calls and trunk group "east17" for outbound dial peer matching:

```
Router(config)# voice source-group alpha
Router(cfg-source-grp)# trunk-group-label source north3
Router(cfg-source-grp)# trunk-group-label target east17
```

Related Commands

Command	Description
carrier-id (dial-peer)	Specifies the carrier associated with a VoIP call.
show voice source-group	Displays the configuration for voice source IP groups.

trustpoint (DSP farm profile)

To associate a trustpoint with a DSP farm profile, use the **trustpoint** command in DSP farm profile configuration configuration mode. To remove the association, use the **no** form of this command.

trustpoint *trustpoint-label*
no trustpoint *trustpoint-label*

Syntax Description

trustpoint-label	Label of the trustpoint to be associated with the digital signal processor (DSP) farm profile.

Command Default

No trustpoints are associated with the DSP farm profile

Command Modes

DSP farm profile configuration (config-dspfarm-profile)

Command History

Release	Modification
12.4(11)XW1	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to associate trustpoints with secure DSP farm profiles only. Use the **security** keyword of the **dspfarm profile** command to configure a secure DSP farm profile. If the trustpoint is not already configured, you are prompted to configure the trustpoint.

Examples

The following example associates the trustpoint dspfarm with the DSP farm profile 101:

```
Router(config)# dspfarm profile 101 conference security
Router(config-dspfarm-profile)# trustpoint dspfarm
```

Related Commands

Command	Description
dspfarm profile	Enters DSP farm profile configuration mode and defines a profile for digital signal processor (DSP) farm services.

trustpoint (voice class)

To configure a trustpoint, and associate it to a TLS profile, use the **trustpoint** command in voice class configuration mode. To delete the trustpoint, use the **no** form of this command.

trustpoint *trustpoint-name* [**client** *client-trustpoint-name*]
no trustpoint

Syntax Description

trustpoint-name	**trustpoint** *trustpoint-name* —creates a trustpoint to store the devices certificate generated as part of the enrollment process using Cisco IOS public-key infrastructure (PKI) commands.
client *client-trustpoint-name*	(Optional) Associate a client trustpoint. Specify the client trustpoint label.

Command Default

No default behavior or values

Command Modes

Voice class configuration (config-class)

Command History

Release	Modification
Cisco IOS XE Amsterdam 17.3.1a	This command was introduced under voice class configuration mode. Introduced support for Yang Model.
Cisco IOS XE 26.1.1	This command was modified to allow provisioning and assigning separate certificates for client and server roles on each SIP trunk in CUBE.

Usage Guidelines

The trustpoint is associated to a TLS profile through the command **voice class tls-profile** *tag*. The *tag* associates the trustpoint configuration to the command **crypto signaling**.

Starting from Cisco IOS XE 26.1.1 release, you can configure client trustpoint along with existing trustpoint configuration under **voice class tls profile**. The client trustpoint is used for TLS client authentication. If client trustpoint is not configured then it'll fallback to common trustpoint .

Examples

The following example illustrates how to create a voice class tls-profile and associate a trustpoint to be used by Cisco UBE to establish a connection with a remote device:

```
Router(config)#voice class tls-profile 2
Router(config-class)#trustpoint CUBETP
```

The following is a sub option to configure client trustpoint along with existing trustpoint config under **voice class tls profile**:

```
Router(config)#voice class tls-profile 1
Router(config-class)#trustpoint  CUBE_TP client Client_TP
```

Related Commands

Command	Description
voice class tls-profile	Provides sub-options to configure the commands that are required for a TLS session.
crypto signaling	Identifies the trustpoint or the **tls-profile** *tag* that is used during the TLS handshake process.

ttl

To set the expiration timer for advertisements, enter the **ttl**command in Annex G configuration mode. To reset to the default, use the no form of this command.

ttl *ttl-value*
no ttl

Syntax Description

ttl -value	Amount of time (in seconds) for which a route from a neighbor is considered valid. Range is from 1 to 2147483647. The default is 1800 (or 30 minutes).

Command Default

1800 seconds (30 minutes)

Command Modes

Annex G configuration

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.

Usage Guidelines

The address templates or routes that are static to this Annex G border element (BE) can be advertised to its neighbors. A time-to-live (TTL) value is associated with each of the advertised routes. The TTL value indicates how long the neighbor should consider the routes valid. On expiration of the ttl, the neighbor must query the addressing information again.

Examples

The following example shows a BE with a time-to-live value of 20 seconds.

```
Router(config)# call-router h323-annexg be20
Router(config-annexg)# ttl 20
```

Related Commands

Command	Description
call -router	Enables the Annex G BE configuration commands.
show call -router status	Displays the Annex G BE status.

type (settlement)

To point to the provider type and the specific settlement server, use the **type** command in settlement configuration mode. To disable this command, use the **no** form of this command.

type {**osp** | **uni-osp**}
no type

Syntax Description

osp	Enables the Open Settlement Protocol (OSP) server type.
uni -osp	Enables authentication of VoIP calls to the Public Switched Telephone Network (PSTN) using a single settlement server.

Command Default

osp

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced on Cisco 2600 series and Cisco 3600 series, and Cisco AS5300.
12.1(2)T	The uni-osp keyword was introduced.

Usage Guidelines

This command defines the settlement server that is doing the accounting and enables the server to do the accounting.

Examples

The following example enables authentication of VoIP calls to the PSTN using a single settlement server:

```
settlement 0
 type uni-osp
```

Related Commands

Command	Description
connection -timeout	Sets the connection timeout.
customer -id	Sets the customer identification.
device -id	Sets the device identification.
encryption	Specifies the encryption method.
max -connection	Sets the maximum simultaneous connections.
response -timeout	Sets the response timeout.
retry -delay	Sets the retry delay.

Command	Description
retry -limit	Sets the connection retry limit.
session -timeout	Sets the session timeout.
settlement	Enters settlement configuration mode.
show settlement	Displays the configuration for all settlement server transactions.
shutdown/no shutdown	Brings up the settlement provider and then shuts it down.
url	Specifies the Internet service provider (ISP) address.

type (voice)

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

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

Syntax Description

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

Command Default

Type 1

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

Use the **type** command to specify the E&M interface for a particular voice port. With **1**, the tie-line equipment generates the E-signal to the PBX type grounding the E-lead. The tie-line equipment detects the M-signal by detecting current flow to ground. If you select **1**, a common ground must exist between the line equipment and the PBX.

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



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

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

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

Examples

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

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