# Cisco IOS Voice Commands: T

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

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

# tbct clear call

To terminate billing statistics for one or more active Two B-Channel Transfer (TBCT) calls, use the **tbct clear call** command in privileged EXEC mode.

**tbct clear call** {**all** | *interface* [*call-tag*]}

## Syntax Description

| | |
|---|---|
| **all** | Active TBCT calls on all interfaces. |
| *interface* | Active TBCT calls on a specified interface. Range is platform-dependent. |
| *call-tag* | (Optional) A specific active TBCT call on the specified interface, as identified by the unique call tag number. Range is 1 to 4,294,967,295. |

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

| Release | Modification |
|---|---|
| 12.3(1) | This command was introduced. |

## Usage Guidelines

- Use this command to manually clear a specific active call or a group of active calls, if, for instance, the ISDN switch goes down. You should not have to manually clear calls with this command unless there is a problem with the switch.
- This command terminates billing information that is being sent to the RADIUS server if, for some reason, the gateway did not receive a notify message from the switch that a call has cleared.
- To automatically clear calls after a specified duration, use the **tbct max call-duration** command.
- To determine the *interface* and *call-tag* arguments to use with this command, use the **show call active voice redirect** command.

## Examples

The following example clears calls on T1 interface 6/0:

```
Router# tbct clear call T1-6/0
```

## Related Commands

| Command | Description |
|---|---|
| **isdn supp-service tbct** | Enables ISDN TBCT on PRI trunks. |
| **show call active voice redirect** | Displays information about active calls that are being redirected using RTPvt or TBCT. |

| Command | Description |
|---|---|
| **tbct max call-duration** | Sets the maximum duration allowed for a call that is redirected using TBCT. |
| **tbct max calls** | Sets the maximum number of active calls that can use TBCT. |

# tbct max call-duration

To set the maximum duration allowed for a call that is redirected using Two B-Channel Transfer (TBCT), use the **tbct max calls** command in global configuration mode. To reset to the default, use the **no** form of this command.

**tbct max call-duration** *minutes*

**no tbct max call-duration**

## Syntax Description

| | |
|---|---|
| *minutes* | Maximum duration, in minutes, allowed for a single TBCT call. Range is 1 to 9999, in recommended increments of 5 minutes. Default is no limit. |

## Defaults

No limit

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(1) | This command was introduced. |

## Usage Guidelines

- Use this command to automatically clear stale calls, for instance if the PRI trunk goes down. To manually clear calls, use the **tbct clear call** command.
- Cisco recommends that you set the call duration in increments of 5 minutes.

**Note** The call duration limit set by this command is not precisely enforced; calls may not be cleared after the exact number of minutes specified by this command.

## Examples

The following example clears TBCT calls that last longer than 10 minutes:

```
tbct max call-duration 10
```

## Related Commands

| Command | Description |
|---|---|
| **isdn supp-service tbct** | Enables ISDN TBCT on PRI trunks. |
| **show call active voice redirect** | Displays information about active calls that are being redirected using RTPvt or TBCT. |
| **tbct clear call** | Terminates billing statistics for one or more active TBCT calls. |
| **tbct max calls** | Sets the maximum number of active calls that can use TBCT. |

# tbct max calls

To set the maximum number of active calls that can use Two B-Channel Transfer (TBCT), use the **tbct max calls** command in global configuration mode. To reset to the default, use the **no** form of this command.

**tbct max calls** *number*

**no tbct max calls**

## Syntax Description

| | |
|---|---|
| *number* | Maximum number of currently active calls that can invoke TBCT at any one time. Range is 1 to 1,000,00. Default is no limit. |

## Defaults

No limit, except as allowed by memory resources

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(1) | This command was introduced. |

## Usage Guidelines

Use this command to control memory resources on the gateway by limiting the amount of memory consumed by TBCT calls.

## Examples

The following example sets the maximum number of calls using TBCT to 500:

```
tbct max calls 500
```

## Related Commands

| Command | Description |
|---|---|
| **isdn supp-service tbct** | Enables ISDN TBCT on PRI trunks. |
| **show call active voice redirect** | Displays information about active calls that are being redirected using RTPvt or TBCT. |
| **tbct clear call** | Terminates billing statistics for one or more active TBCT calls. |
| **tbct max call-duration** | Sets the maximum duration allowed for a call that is redirected using TBCT. |

# tdm-group

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

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

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

## Syntax Description

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

## Defaults

No TDM group is configured.

## Command Modes

Controller configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)MA | This command was introduced on Cisco MC38310. |
| 12.1(1)T | This command was modified to include voice WAN interface cards (VWICs) for Cisco 2600 series and Cisco 3600 series. |
| 12.1(2)T | This command was modified for the OC-3/STM-1 ATM Circuit Emulation Service network module on Cisco 2600 series and Cisco 3600 series. |

## Usage Guidelines

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

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

## Examples

The following example configures TDM group 1 to include timeslots 13 through 20:

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

The following example configures TDM group number 20 on controller T1 1 to support Foreign Exchange Office (FXO) ground-start:

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

## Related Commands

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

# tech-prefix

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

**tech-prefix** *number*

**no tech-prefix**

## Syntax Description

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

## Defaults

No technology prefix is defined.

## Command Modes

Dial-peer configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(6)NA2 | This command was introduced on Cisco 2600 series and Cisco 3600 series. |
| 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

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

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

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

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

## Examples

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

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

## Related Commands

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

# telephony-service

To enable Cisco IOS Telephony Service and enter telephony-service configuration mode, use the **telephony-service** command in global configuration mode. To disable Cisco IOS Telephony Service, use the **no** form of this command.

**telephony-service**

**no telephony-service**

## Syntax Description

This command has no arguments or keywords.

## Defaults

Cisco IOS Telephony Service is disabled by default.

## Command Modes

Global configuration

## Command History

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

## Usage Guidelines

The **telephony-service** command is the top-level command for all other commands related to Cisco IOS Telephony Service configuration.

## Examples

The following example enters telephony-service configuration mode:

```
Router(config)# telephony-service
Router(config-telephony-service)#
```

## Related Commands

| Command | Description |
|---|---|
| **dialplan-pattern** | Creates a global prefix that can be used to expand the abbreviated extension numbers into fully qualified E.164 numbers. |
| **ephone** | Enters ephone configuration mode. |
| **ephone-dn** | Enters ephone-dn configuration mode. |
| **ip source-address** | Identifies the IP address and port number that the Cisco IOS Telephony Service router uses for the IP phone service. |

| Command | Description |
|---|---|
| **keepalive** | Configures the time interval between sending keepalive messages to the router used by the Cisco IP phones. |
| **load** | Identifies the Cisco IP phone firmware to be used by the Cisco IP phone. |
| **max-dn** | Configures maximum number of directory numbers supported by the Cisco IOS Telephony Service router. |
| **max-ephones** | Configures the maximum number of Cisco IP phones supported by the Cisco IOS Telephony Service router. |
| **reset** | Resets the Cisco IP phone. |
| **timeouts interdigit** | Configures the interdigit timeout value for all Cisco IP phones attached to the router. |
| **transfer-pattern** | Allows transfer of telephone calls to other non-IP phone numbers. |
| **url** | Provisions URLS for use by the Cisco IP phones connected to the Cisco IOS Telephony Service router. |
| **voicemail** | Configures the telephone number that is speed-dialed when the message button on a Cisco IP phone is pressed. |

# test call fallback probe

To test current network conditions against a particular IP address and to display the Calculated Planning Impairment Factor (ICPIF) Service Assurance Agent (SAA) values, use the **test call fallback probe** command in EXEC mode.

**test call fallback probe** *ip-address* [**codec** {**711** | **729**}]

## Syntax Description

| | |
|---|---|
| *ip-address* | Target IP address. |
| **codec** | (Optional) Codec type to test. The keywords are as follows:<br>• **711**—G.711 codec<br>• **729**—G.729 codec |

## Command Modes

EXEC

## Command History

| Release | Modification |
|---|---|
| 12.1(3)T | This command was introduced on Cisco 2600 series, Cisco 3600 series, and Cisco MC3810. |
| 12.2(2)XA | The **call fallback** and **call fallback reject-cause-code** commands were introduced. |
| 12.2(2)XB1 | This command was implemented on the Cisco AS5850. |
| 12.2(4)T | The PSTN Fallback feature and enhancements were introduced on Cisco 7200 series routers and integrated into Cisco IOS Release 12.2(4)T. |
| 12.2(4)T2 | This command was implemented on Cisco 7500 series routers. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T. |

## Usage Guidelines

Use the **test call fallback probe** command to get an immediate look at current network conditions in terms of the ICPIF value between your router and a specific destination. This command is used for statistic collection. You may determine that further action needs to be taken upon receiving information.

This command has no impact on the cache.

## Examples

The following example demonstrates a test probe to IP address 10.0.0.0 and shows that the ICPIF value to 10.0.0.0 is 0. No network congestion currently exists.

```
Router# test call fallback probe 10.0.0.0

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

## Related Commands

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

# test call id

To manipulate the echo canceller and jitter buffer parameters in real time, use the **test call id** command in privileged EXEC mode.

**test call id** *call-id* {**echo-canceller**{**coverage** *range-in-ms* | **erl worst-case** {**0** | **3** | **6**}| **h-register** {**clear** | **freeze** | **thaw**}} | **playout-delay** {**fixed** | **adaptive** {*nominal-delay min-delay max-delay*}}}

## Syntax Description

| | |
|---|---|
| *call-id* | The hexadecimal ID of an active voice call. Values can be from 0 to FFFFFFFF. |
| **echo-canceller** | Tests the echo canceller on an active voice call. |
| **coverage** *range-in-ms* | Tests echo canceller coverage in milliseconds. Valid values are 0, 8, 16, 24, 32, 48, 64, and 128, where 64 is the default value for the extended EC and 8 is the default value for NextPort firmware. Specific default values depend on which echo canceller and firmware you are using:<br>• Standard echo canceller (Cisco-proprietary G.165 EC)—8 ms<br>• Extended echo canceller—64 ms<br>• NextPort firmware—8 ms<br>See the “Usage Guidelines” section for more information about default values. |
| **erl worst-case** {**0** \| **3** \| **6**} | Determines worst-case Echo Return Loss (ERL), in decibels (dB). Values can be 0, 3, or 6. Default is 6.<br>**Note** The **echo-canceller erl worst-case** keywords combine to form a tunable parameter available with the extended echo canceller only. The **erl** option is available only with the extended echo canceller. |
| **h-register** | Controls the extended echo canceller h-register. |
| {**clear** \| **freeze** \| **thaw**} | Clears, freezes, or thaws a call in the extended echo canceller h-register. |
| **playout-delay** | Resets the playout buffering on the associated digital signal processors (DSPs) to the requested values. If **fixed** *fixed-delay* is selected, the first parameter only is required and used. If all three parameters are used, they are accepted, but the last two are ignored. If **adaptive** *nominal-delay min-delay max-delay* is selected, all three values are required and used. |
| **fixed** *fixed-delay* | Tests the fixed playout-delay mode. Jitter buffer size does not adjust during a call; a constant playout delay is added. The *fixed-delay* argument is nominal delay in ms. Range is from 0 to 1500. |
| **adaptive** *nominal-delay min-delay max-delay* | Tests the adaptive playout-delay mode. Adjusts jitter buffer size and amount of playout delay during a call on the basis of current network conditions. If the **adaptive** keyword is used, *nominal-delay*, *min-delay*, and *max-delay* are sanity checked for maximum delay being greater than or equal to the nominal delay, which is greater than or equal to the minimum delay.<br>Nominal delay range is from 0 to 1500 ms. Minimum delay range is from 10 to 80 ms. Maximum delay range is from 40 to 1700 ms.<br>**Note** These options cause audible disturbance to the call and should be used with care. |

## Command Modes

Privileged EXEC

## Command History

| Release | Modification |
|---|---|
| 12.2(13)T | This command was introduced on all voice platforms with echo cancellation and extended echo cancellation. |

## Usage Guidelines

The *call-id* argument value must be obtained by using the **show voice call status** command. The following is an example of how to obtain the call ID. The first parameter displayed in the output of the **show voice call status** command is the call ID.

**Note** You should not use the "0x" prefix in the *call-id* argument when you enter the resulting call ID in the **show voice call status** command.

```
Router# show voice call status

CallID     CID  ccVdb        Port      DSP/Ch  Called #   Codec    Dial-peers
0x2        11D1 0x62FE6478 1/0/0     1/1     10001      g711ulaw 1/2
0x3        11D1 0x62FE80F0 1/0/1     2/1    *10001      g711ulaw 2/1
1 active call found
```

The **echo-cancel coverage** ranges can differ depending on the platform and DSP code configuration. See Table 158 for the echo canceller coverage ranges.

Some of the options in the Syntax Description table can be used only on specific platforms that are running the extended echo canceller. Table 158 lists the platforms supported with this feature and whether the standard (TI C54x voice-based platforms) or the extended (NextPort/Conexant voice-based platforms) echo canceller is available on that platform. A disabled state is indicated by 0.

***Table 158 Echo Canceller Types and Canceller Coverage Ranges***

| Platform | Echo Canceller Type | Echo Canceller Coverage Range |
|---|---|---|
| Cisco 827 | Standard | Standard—0, 8, 16, 24, 32 |
| Cisco 1700 series | Standard | Standard—0, 8, 16, 24, 32 |
| Cisco 2400 series | Standard | Standard—0, 8, 16, 24, 32 |
| Cisco 2600 series | Standard, Extended | Standard—0, 8, 16, 24, 32<br>Extended—0, 24, 32, 48, 64 |
| Cisco 3600 series | Standard, Extended | Standard—0, 8, 16, 24, 32<br>Extended—0, 24, 32, 48, 64 |
| Cisco 7200 series | Standard, Extended | Standard—0, 8, 16, 24, 32<br>Extended—0, 24, 32, 48, 64 |
| Cisco 7750 | Standard | Standard—0, 8, 16, 24, 32 |
| Cisco AS5300 | Standard, Extended | Standard—0, 8, 16, 24, 32<br>Extended—0, 24, 32, 48, 64 |
| Cisco AS5350 | NextPort | NextPort—0, 8, 16, 24, 32, 64, 128 |

*Table 158 Echo Canceller Types and Canceller Coverage Ranges (continued)*

| Platform | Echo Canceller Type | Echo Canceller Coverage Range |
|---|---|---|
| Cisco AS5400 | NextPort | NextPort—0, 8, 16, 24, 32, 64, 128 |
| Cisco AS5800 | Standard | Standard—0, 8, 16, 24, 32 |
| Cisco AS5850 | NextPort | NextPort—0, 8, 16, 24, 32, 64, 128 |
| Cisco CVA122 | Standard | Standard—0, 8, 16, 24, 32 |
| Cisco MC3810 | Standard, Extended | Standard—0, 8, 16, 24, 32<br>Extended—0, 24, 32, 48, 64 |
| Cisco uBR92x | Standard | Standard—0, 8, 16, 24, 32 |

**Note** The keywords and arguments in the Syntax Description table requests that the specified parameters be sent to the DSP using the normal DSP control message mechanism expecting an immediate effect. You can expect a short discontinuity and settling period for the voice stream. These parameters have effect only for the duration of the call. Echo-canceller and playout parameters revert to the values defined in the configuration on the next call using that DSP.

You can use this command with the extended echo canceller, which allows you to configure the voice card in a router individually, or with the standard echo canceller, in which the configuration occurs implicitly on the router. The following two new output messages are possible with the extended echo cancellation feature when either an extended-only or a standard-only echo cancellation function is requested:

```
Extended echo canceller not active for CallID callID
Basic echo canceller not active for CallID callID
```

The CLI help strings typically show which version of echo canceller is running and if it is valid for the requested function. For example:

```
Router# test call id 3 echo-canceller erl worst-case ?

  0  worst case extended echo canceller operation at 0 dB ERL
  3  worst case extended echo canceller operation at 3 dB ERL
  6  worst case extended echo canceller operation at 6 dB ER

Router# test call id 3 echo-canceller coverage ?

  0   disable echo-canceller
  16  16 ms echo canceller coverage (basic only)
  24  24 ms echo canceller coverage (basic & extended)
  32  32 ms echo canceller coverage (basic & extended)
  48  48 ms echo canceller coverage (extended only)
  64  64 ms echo canceller coverage (extended only)
  8   8 ms  echo canceller coverage (basic only)
```

In its section on testing echo cancellers, ITU-T specification G.168 invents a hypothetical device in the EC called an h-register. The h-register stores the impulse response of the echo path and invents actions such as “clear the h-register,” “contents of the h-register are frozen,” and “thaw” to undo the “freeze.” The h-register is the filter within EC used to estimate the echo. If it freezes, its filter coefficients do not adapt to the signal. If there is a significant change in the signal characteristic, such as power level or delay, echo is heard.

The h-register test mode settings allow manual manipulation of the EC h-register for G.168-like tests. Actual G.168 testing is embedded in the digital signal processor (DSP) and does not require explicit Cisco IOS control of the h-register. The call ID must be a valid active telephony call leg ID as displayed by entering the **show call active brief** command in privileged EXEC mode.

## Examples

The following example shows how to experiment in real time with the parameters of an active call. In this example, the nominal delay for both the **adaptive** and **fixed** options is 5 ms; the minimum delay for the **adaptive** option is 10 ms; and the maximum delay for the **adaptive** option is 40 ms.

```
Router# test call id 99 playout-delay fixed 5

Router# test call id 99 playout-delay adaptive 5 10 40
```

The following example shows the echo canceller range on the associated DSPs being reset:

```
Router# test call id 99 echo-canceller coverage 0
```

The following example show the **test call id** command clearing the h-register in the extended echo canceller:

```
Router# test call id 02 echo-canceller h-register clear
```

## Related Commands

| Command | Description |
|---|---|
| **show call threshold** | Displays enabled triggers, current values for configured triggers, and number of API calls that were made to global and interface resources. |
| **show call treatment** | Displays the call treatment configuration and the statistics for handling the calls based upon resource availability. |
| **show voice call** | Shows the real-time call status for voice ports on the Cisco router or concentrator. |
| **show call active** | Displays active call information for voice calls or fax transmissions in progress. |
| **test call fallback probe** | Tests current network conditions to a particular IP address and displays the ICPIFSAA values. |
| **test call threshold** | Tests how the core APIs behave on the basis of the resource configuration. |

# test call threshold

To test how the core application programming interfaces (APIs) behave with respect to the resource configuration, use the **test call threshold** command in privileged EXEC mode.

**test call threshold** {**enable** [**busyout** | **treatment**] [**global** | **ipaddress** *ipaddress*] | **interface** *interface-name interface-number*}

## Syntax Description

| | |
|---|---|
| **enable** | Enables busyout or treatment action. Default is both. |
| **busyout** | Autobusyouts the T1 or E1 if the resource is not available. |
| **treatment** | Applies call treatment from session application if the resource is not available. |
| **global** | Test to be on the global resources on gateway. |
| **ipaddress** *ipaddress* | Remote address. Allows users to test how the core behaves. |
| **interface** *interface-name interface-number* | Interface that is configured as a gateway. |

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XA | This command was introduced. |
| 12.2(4)T | The 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 the Cisco AS5850 universal gateway. |
| 12.2(4)XM | This command was implemented on Cisco 1750 and Cisco 1751 routers. Support for other Cisco platforms is not included 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 on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release. |
| 12.2(11)T | This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release. |

## Examples

The following example tests the global resources.

```
test call threshold enable global
```

## Related Commands

| Command | Description |
|---|---|
| **call spike** | Limits the number of calls that can be received from the PSTN in a configured time period. |
| **call threshold poll-interval** | Enables a polling interval threshold for CPU or memory. |
| **show call threshold** | Displays enabled triggers, current values for configured triggers, and number of API calls that were made to global and interface resources. |
| **show call treatment** | Displays the call treatment configuration and statistics for handling calls on the basis of availability. |

# test enum

To test the functionality of an ENUM match table, use the **test enum** command in privileged EXEC mode.

**test enum** *table-number input-pattern*

## Syntax Description

| | |
|---|---|
| *table-number* | Number of the ENUM match table to be tested. Range is from 1 to 15. |
| *input-pattern* | Stream editor (SED) expression to be resolved using the ENUM match table. |

## Defaults

No default behavior or values.

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

The **test enum** command emulates an ENUM call with the input pattern as the called number. The **contact_list** field in the display contains the URLs returned by the ENUM server.

## Examples

Given the following definition for ENUM match-table 3:

```
voice enum_match_table 3
rule 1 5 /^9\(1.*\)/ /+\1/ cisco
rule 2 4 /^9011\(.*\)/ /+1408\1/ arpa
rule 10 1 /^(.*)/ /\1/ e164.cisco.com
```

the following example tests the input string 12345 against ENUM match table 3:

```
Router# test enum 3 12345

tahoe3#contact_list :
              tel:1111
contact_list :
              sip:345789@contact1.alpha.com
contact_list :
              tel:765
contact_list :
              sip:12345@contact1.alpha.com
contact_list :
              sip:987@contact2.alpha.com
contact_list :
              h323:12345@contact2.alpha.com:5060
contact_list :
              h323:12345@contact1.alpha.com:5060
```

```
contact_list :
              h323:12345@contact3.alpha.com:5060
contact_list :
              sip:654@172.18.188.173
contact_list :
              tel:876
enum_test_command: contact_list 62E4E8A4
```

## Related Commands

| Command | Description |
|---|---|
| **rule (ENUM configuration)** | Defines the match and replace patterns for the ENUM rule. |
| **show voice enum-match-table** | Displays the configuration for voice ENUM match tables. |
| **voice enum-match-table** | Initiates the definition of an ENUM match table. |

# test pots dial

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

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

## Syntax Description

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

## Command Modes

EXEC

## Command History

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

## Usage Guidelines

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

## Examples

The following POTS dial command dials the telephone number 4085551234:

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

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

## Related Commands

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

# test pots disconnect

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

**test pots** *port* **disconnect**

## Syntax Description

| | |
|---|---|
| *port* | Port number 1 or 2. |

## Command Modes

EXEC

## Command History

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

## Examples

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

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

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

## Related Commands

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

# test source-group

To test the functionality of a source group, use the **test source-group** command in privileged EXEC mode.

**test source-group** {**carrier-id source** *name* | **h323zone-id** *name* | **ip-address** *ip-address* | **trunk-group-label source** *name*}

## Syntax Description

| | |
|---|---|
| **carrier-id source** *name* | Source carrier ID of the source group to be tested. |
| **h32zone-id** *name* | Name of the H.323 zone source group to be tested. |
| **ip-address** *ip-address* | IP address of the source group to be tested. |
| **trunk-group-label source** *name* | Trunk group label of the source group to be tested. |

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

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

## Examples

The following example tests the source group with carrier ID “newyork”:

```
Router# test source-group carrier-id source newyork

A source-group is found with source carrier-id newyork


Source Group: 130
    description=””,
    carrier-id source=”newyork”,
    carrier-id target=”dallas”,
    trunk-group-label source=””,
    trunk-group-label target=””,
    h323zone-is=””,
    access-list=,
    disconnect-cause=”no-service”
    translation-profile=”abc-profile-sipg”,
Router#
```

The following example tests the source group with H.323 zone ID “sanjose”:

```
Router# test source-group h323zone-id sanjose

A source-group is found with h323zone-id sanjose


Source Group: ma101
    description=””,
```

```
    carrier-id source="",
    carrier-id target="",
    trunk-group-label source="",
    trunk-group-label target="",
    h323zone-is="sanjose",
    access-list=,
    disconnect-cause="no-service"
    translation-profile="abc-profile-sipg",
```

The following example tests the source group using an IP address:

```
Router# test source-group ip-address 172.16.100.100

A source-group is found with ip-address=172.16.100.100


Source Group: 110
    description="",
    carrier-id source="",
    carrier-id target="",
    trunk-group-label source="",
    trunk-group-label target="",
    h323zone-is="",
    access-list=,
    disconnect-cause="no-service"
    translation-profile="abc-profile-sipg",
```

The following example tests the source group with the trunk-group label "losangeles":

```
Router# test source-group trunk-group-label source losangeles

A source-group is found with source trunk-group-label losangeles


Source Group: 130
    description="",
    carrier-id source=""'
    carrier-id target=""'
    trunk-group-label source="losangeles",
    trunk-group-label target="chicago",
    h323zone-is="",
    access-list=,
    disconnect-cause="no-service"
    translation-profile="abc-profile-sipg",
```

The following example displays the error message for a nonexistent carrier ID or trunk-group label:

```
Router# test source-group carrier-id source 1511

No source-group is found with input source carrier-id 1511
```

## Related Commands

| Command | Description |
|---|---|
| **show voice source-group** | Displays the configuration for voice source IP groups. |
| **voice source-group** | Initiates the voice source-group definition. |

# test translation-rule

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

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

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

## Syntax Description

| | |
|---|---|
| *name-tag* | The tag number by which the rule set is referenced. This is an arbitrarily chosen number. Range is from 1 to 2147483647. |
| *input-number* | The input string of digits for which a pattern matching is performed. |
| *input-numbering-type* | (Optional) The keyword choices for this field are **international**, **national**, **subscriber**, **abbreviated**, **unknown**, and **any**. |

## Defaults

No default behavior or values

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.0(7)XR1 | This command was introduced for VoIP on Cisco AS5300. |
| 12.0(7)XK | This command was implemented for the following voice technologies on the following platforms:<br>• VoIP (Cisco 2600, Cisco 3600, and Cisco MC3810)<br>• Voice over Frame Relay (Cisco 2600, Cisco 3600, and Cisco MC3810)<br>• Voice over ATM (Cisco 3600 and Cisco MC3810) |
| 12.1(1)T | This command was integrated into Cisco IOS Release 12.1(1)T for the following voice technology on the following platforms:<br>• VoIP (Cisco 1750, Cisco 2600, Cisco 3600, Cisco AS5300, Cisco 7200, and Cisco 7500) |
| 12.1(2)T | This command was implemented on the T train for the following voice technologies on the following platforms:<br>• VoIP (Cisco MC3810)<br>• Voice over Frame Relay (Cisco 2600, Cisco 3600, and Cisco MC3810)<br>• Voice over ATM (Cisco 3600 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. |

## Examples

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

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

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

## Related Commands

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

# test voice echo

To check echo cancellation on a port, use the **test voice echo** command in global configuration mode.

### Cisco Modular Access Routers with Analog Voice Ports

**test voice echo** *slot*/*subunit*/*port echo-encapsulation-bit-mask*

### Cisco Modular Access Routers with Digital Voice Ports

**test voice echo** *slot*/*port*:*ds0-group echo-encapsulation-bit-mask*

## Syntax Description

**For Cisco Modular Access Routers with Analog Voice Ports:**

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

**For Cisco Modular Access Routers with Digital Voice Ports:**

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

**All Platforms:**

| | |
|---|---|
| *echo-encapsulation -bit-mask* | Sets the DSP firmware echo canceller control. This is a 16-bit hex string. Valid ranges are from 0 to 0x0FFF. See Table 159 for valid bit settings. |

*Table 159* **echo_encapsulation_bit_mask** ***Bit Settings***

| Bit | Subparameter | Settings | Description | DSPWare Version |
|---|---|---|---|---|
| 0 | ec_disable | Echo canceller disable control:<br>0=enable (default)<br>1=disable | Echo cancellation can be disabled through the ec_disable subparameter. When the echo canceller is disabled, the echo filter and all its internal state is kept but not updated. The input signal does not get processed by the echo canceller filter when disabled. | All |
| 1 | ec_freeze | echo canceller freeze control:<br>0=echo filter update enable (default)<br>1=freeze echo filter (disable update) | The echo canceller filter can be frozen using the ec_freeze subparameter. While the echo canceller is frozen, the echo canceller filter and all other internal adaptive states are not updated. | All |
| 2 | nlp_disable | NLP disable control:<br>0=enable NLP (default)<br>1=disable NLP | The Non-Linear Processor (NLP) can be disabled through the nlp_disable subparameter. The NLP is used to suppress residual echo (either replacing the residual echo with silence or with background noise). If disabled, residual echo will not be suppressed. The NLP, if enabled, can be configured to replace the suppressed residual echo with either silence or comfort noise using the cn_disable subparameter. If comfort noise is disabled, the NLP will replace suppressed residual with silence. The NLP activation threshold can be modified using the threshold optional parameter. The activation threshold is used in determining at what level of residual echo (relative to the far end signal) the NLP will activate. The NLP activation has a significant affect on the subjective quality of the echo canceller in double-talk, high background noise, or high tail length situations. If threshold is not sent in the *echo_encapsulation_bit_mask* argument, a value of –21 dB is used. Threshold only applies to the standard echo canceller. The extended echo canceller does not have a configurable NLP threshold. | All |
| 3 | ec_reset | Echo canceller reset control:<br>0=no reset<br>1=reset echo canceller filter (default) | The ec_reset subparameter will reset the echo canceller filter and all internal adaptive state of the echo canceller. If the echo suppressor is enabled, it will also be reset and activate for the configured coverage time. ec_reset subparameter differs from the other subparameters in that it is a trigger flag. The internal state of ec_reset will be set to zero (0) after executing the echo reset. | All |

**Table 159 echo_encapsulation_bit_mask Bit Settings (continued)**

| Bit | Subparameter | Settings | Description | DSPWare Version |
|---|---|---|---|---|
| 4 | hpf_disable | High pass filter disable control:<br>0= enable (default)<br>1=disable | The echo canceller is preceded by a high pass filter that operates on the input (near-end) signal. High pass filtering is required for proper operation of the adaptive echo canceller algorithm. The high pass filter can be disabled by using the hpf_disable subparameter. The high pass filter should not be disabled while the echo canceller filter is operating. This can result in instability of the echo canceller filter updates. For channels that do not want the echo canceller to modify the input signal in any way (e.g. clear-channel applications), both the echo canceller and the high pass filter should be disabled. | All |
| 5 | cn_disable | NLP comfort noise disable:<br>0=enable (default)<br>1=disable | The cn_disable subparameter controls the application of comfort noise. | All |
| 6-7 | worst_erl[1] | Worst case ERL configuration:<br>0=6 dB (default)<br>1=3 dB<br>2=0 dB<br>3=reserved | If the extended echo canceller is selected, the worst_erl subparameter is used. If the standard echo canceller is selected, worst_erl is not used and must be set to zero (0). The worst_erl subparameter configures the lowest amount of Echo Return Loss (ERL) that the echo canceller needs to operate properly. For ERLs lower than worst_erl, the echo canceller does not converge properly. The worst_erl setting does impact the performance of the double-talk detector that is used for much of the operation of the echo canceller, so for best performance it should not be set lower than is required. | 4.1 and later |
| 8-10 | tail_length[1] | Echo tail length<br>0=24 ms<br>1=32 ms<br>2=48 ms<br>3=64 ms (default)<br>4=80 ms<br>5=96 ms<br>6=112 ms<br>7=128 ms | If the extended echo canceller is selected, the tail_length subparameter is used. If the standard echo canceller is selected, the tail_length is not used and must be set to zero (0). The tail length of the standard echo canceller is not configured in the *echo_encapsulation_bit_mask* argument. The tail_length subparameter configures the maximum tail length for the extended echo canceller. Many DSP firmware builds limit the maximum tail length less than 128 ms. If the maximum tail length is limited by the DSP builds, configured values greater than that supported will be set to the maximum supported by the build. | 4.1 and later |

*Table 159* **echo_encapsulation_bit_mask** *Bit Settings (continued)*

| Bit | Subparameter | Settings | Description | DSPWare Version |
|---|---|---|---|---|
| 11 | ecan_type | Select active echo canceller type:<br>0= standard (default)<br>1=extended | For those DSP firmware releases that support both standard and extended echo cancellers, the selection of the type of echo canceller to use is controlled with the ecan_type subparameter. Only a single type of echo canceller can be used for all channels in the DSP. If the nondefault echo canceller type is desired, an *echo_encapsulation_bit_mask* argument with the ecan_type set to the nondefault echo canceller type must be sent to a voice channel before any voice channel is place in a nonIDLE mode. All *echo_encapsulation_bit_mask* arguments sent to the DSP must have the identical setting of ecan_type. There is no changing the echo canceller type in the DSP. Switching between echo canceller types requires a reset of the DSP. | 4.1 and later |
| 12 | sup_enable[2] | Echo suppressor enable control:<br>0=disable (default)<br>1=enable | In addition to the echo canceller, an echo suppressor is implemented in the DSP channel. If enabled through the sup_enable subparameter, the echo suppressor is activated when the echo canceller is reset, either due to a mode change (for example, from IDLE to VOICE), because of a bearer channel detects an echo suppression enable, or because of a echo canceller reset initiated by an *echo_encapsulation_bit_mask* argument. The echo suppressor will remain enabled for seven seconds. The echo suppressor is only available when running the standard echo canceller. All configuration information for the echo suppressor is ignored when running the extended echo canceller. | All |
| 13-15 | reserved | Reserved bits (must be set to zero) | — | All |

1. Subparameter only valid for extended echo canceller. Must be set to zero for standard echo canceller.
2. Subparameter only valid for standard echo canceller. Must be set to zero for extended echo canceller.

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

| Release | Modification |
|---|---|
| 12.0(7)XK | This command was introduced. |

## Usage Guidelines

A call must be in progress in order for this command to work.

The *echo_encapsulation_bit_mask* argument selects the type of echo canceller to use (standard or extended) and provides control for the voice channel's selected echo canceller. The echo canceller control information is kept statically in the voice channel for all modes. When the voice channel transitions from a mode that does not use the echo canceller (such as fax) to a mode that uses the echo canceller (such as voice), the static echo control information is used. The static echo control information is modified by using the *echo_encapsulation_bit_mask* argument.

## Related Commands

| Command | Description |
|---|---|
| **echo-cancel coverage** | Specifies the amount of coverage for echo cancellation. |
| **echo-cancel enable** | Enables echo cancellation on a voice port. |
| **echo suppressor** | Enables echo suppression to reduce initial echo before the echo canceller converges. |
| **non-linear** | Enables nonlinear processing in the echo canceller. |

# test voice playout

To test the playout buffer which accommodates packet jitter caused by switches in the WAN, use the **test voice playout** command in privileged EXEC mode.

### Cisco Modular Access Routers with Analog Voice Ports

**test voice playout** {**adaptive** | **fixed** | **nots**} *initial-size minimum-size maximum-size fax-nominal-size slot*/*subunit*/*port*

### Cisco Modular Access Routers with Digital Voice Ports

**test voice playout** {**adaptive** | **fixed** | **nots**} *initial-size minimum-size maximum-size fax-nominal-size slot*/*port***:***ds0-group*

## Syntax Description

### All Platforms

| | |
|---|---|
| **adaptive** | Identifies an adaptive playout buffer. Jitter buffer size and amount of playout delay are adjusted during a call, on the basis of current network conditions. |
| **fixed** | Identifies a fixed playout buffer. Jitter buffer size does not adjust during a call; a constant playout delay is added. |
| **nots** | Identifies a fixed playout buffer with no timestamps. |
| *initial-size* | Specifies the initial playout buffer size. Range is 0 to 65535. |
| *minimum-size* | Specifies the minimum playout buffer size. Range is 0 to 65535. |
| *maximum-size* | Specifies the maximum playout buffer size. Range is 0 to 65535. |
| *fax-nominal-size* | Specifies the fax nominal playout buffer size. Range is 0 to 65535. |

### For Cisco Modular Access Routers with Analog Voice Ports:

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

### For Cisco Modular Access Routers with Digital Voice Ports:

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

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

| Release | Modification |
|---|---|
| 12.0(7)XK | This command was introduced. |

## Usage Guidelines

A call must be in progress in order for this command to work.

## Related Commands

| Command | Description |
|---|---|
| **playout-delay (dial-peer)** | Tunes the playout buffer on DSPs to accommodate packet jitter caused by switches in the WAN. |
| **playout-delay (voice-port)** | Tunes the playout buffer to accommodate packet jitter caused by switches in the WAN. |
| **playout-delay mode** | Selects fixed or adaptive mode for the jitter buffer on DSPs. |
| **show call active voice** | Displays active call information for voice calls. |

# test voice port detector

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

### Cisco 2600 and Cisco 3600 Series with Analog Voice Ports

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

### Cisco 2600 and Cisco 3600 Series with Digital Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

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

## Syntax Description

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

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

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

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

**For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:**

| | |
|---|---|
| *slot*/*port* | Tests the voice port that you specify with the *slot*/*port* designation.<br>• *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810.<br>• *port* specifies an analog voice port number. Range is from 1 to 6. |

**For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:**

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

**For All Platforms:**

| | |
|---|---|
| **m-lead** | Forces the E&M m-lead detector to the specified state. |
| **loop** | Forces the Foreign Exchange Office (FXO) loop detector to the specified state. |
| **battery-reversal** | Forces the FXO battery-reversal detector to the specified state. |
| **ring** | Forces the FXO ringing detector to the specified state. |
| **tip-ground** | Forces the FXO tip-ground detector to the specified state. |
| **ring-ground** | Forces the Foreign Exchange Station (FXS) ring-ground detector to the specified state. |
| **ring-trip** | Forces the FXS ring-trip detector to the specified state. |
| **on** | Forces the selected item to the on state. |
| **off** | Forces the selected item to the off state. |
| **disable** | Ends the forced state for the selected item. |

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

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

## Examples

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

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

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

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

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

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

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

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

## Related Commands

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

# test voice port inject-tone

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

### Cisco 2600 and Cisco 3600 Series with Analog Voice Ports

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

### Cisco 2600 and Cisco 3600 Series with Digital Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

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

## Syntax Description

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

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

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

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

**For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:**

| | |
|---|---|
| *slot*/*port* | Tests the voice port that you specify with the *slot*/*port* designation.<br>• *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810.<br>• *port* specifies an analog voice port number. Range is from 1 to 6. |

**For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:**

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

**For All platforms:**

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

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

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

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

## Examples

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

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

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

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

or

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

## Related Commands

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

# test voice port loopback

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

### Cisco 2600 and Cisco 3600 Series with Analog Voice Ports

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

### Cisco 2600 and Cisco 3600 Series with Digital Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

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

## Syntax Description

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

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

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

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

**For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:**

| | |
|---|---|
| *slot*/*port* | Tests the voice port that you specify with the *slot*/*port* designation.<br>• *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810.<br>• *port* specifies an analog voice port number. Range is from 1 to 6. |

**For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:**

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

**All Platforms:**

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

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

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

## Examples

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

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

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

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

## Related Commands

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

# test voice port relay

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

### Cisco 2600 and Cisco 3600 Series with Analog Voice Ports

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

### Cisco 2600 and Cisco 3600 Series with Digital Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

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

## Syntax Description

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

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

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

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

**For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:**

| | |
|---|---|
| *slot*/*port* | Tests the voice port that you specify with the *slot*/*port* designation.<br>• *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810.<br>• *port* specifies an analog voice port number. Range is from 1 to 6. |

**For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:**

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

**All Platforms:**

| | |
|---|---|
| **e-lead** | Forces the E&M e-lead relay to the specified state. |
| **loop** | Forces theForeign Exchange Office (FXO) loop relay to the specified state. |
| **ring-ground** | Forces the FXO ring-ground relay to the specified state. |
| **battery-reversal** | Forces the FXO battery-reversal relay to the specified state. |
| **power-denial** | Forces the Foreign Exchange Station (FXS) power-denial relay to the specified state. |
| **ring** | Forces the FXS ringing relay to the specified state. |
| **tip-ground** | Forces the FXS tip-ground relay to the specified state. |
| **on** | Forces the selected item to the on state. |
| **off** | Forces the selected item to the off state. |
| **disable** | Ends the forced state for the selected item. |

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

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

## Examples

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

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

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

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

## Related Commands

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

# test voice port switch

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

### Cisco 2600 and  3600 Series with Analog Voice Ports

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

### Cisco 2600 andCisco  3600 Series with Digital Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Analog Voice Ports

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

### Cisco MC3810 Multiservice Concentrator with Digital Voice Ports

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

## Syntax Description

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

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

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

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

**For the Cisco MC3810 Multiservice Concentrator with Analog Voice Ports:**

| | |
|---|---|
| *slot*/*port* | Tests the voice port that you specify with the *slot*/*port* designation.<br>• *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810.<br>• *port* specifies an analog voice port number. Range is from 1 to 6. |

**For the Cisco MC3810 Multiservice Concentrator with Digital Voice Ports:**

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

**For All Platforms:**

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

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

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

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

## Examples

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

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

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

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

## Related Commands

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

# test voice tone

To test a variety of telephony tones, use the **test voice tone** command in privileged EXEC mode.

### For Tone Testing

**test voice tone** *locale* {**busy** | **congestion** | **dialtone** | **number_unavailable** | **offhook_alert** | **ring_back**} *frequency-number 1st-frequency 2nd-frequency 1st-amplitude-fxs 1st-amplitude-fxo 1st-amplitude-e&m 2nd-amplitude-fxs 2nd-amplitude-fxo 2nd-amplitude-e&m 1st-on 1st-off 2nd-on 2nd-off 3rd-on 3rd-off 4th-on 4th-off*

### For DTMF Testing

**test voice tone** *locale* **dtmf** *r0 r1 r2 r3 c0 c1 c2 c3 a1 a2*

### For Pulse Ratio Testing

**test voice tone** *locale* **pulse** *percent*

### To Show Test Details

**test voice tone** *locale* **show**

## Syntax Description

### All Tests

| | |
|---|---|
| *locale* | Identifies the country with a two-letter ISO-3166 country code shown in Table 160. |

### For Tone Testing

| | |
|---|---|
| **busy** | Specifies testing of the busy tone. |
| **congestion** | Specifies testing of the congestion tone. |
| **dialtone** | Specifies testing of the dial tone. |
| **number_unavailable** | Specifies testing of the tone that occurs when a number is unavailable. |
| **offhook_alert** | Specifies testing of the offhook alert tone. |
| **ring_back** | Specifies testing of the ring-back tone. |
| *frequency-number* | Identifies the number of frequencies to be tested. Valid entries are 1 or 2. |
| *1st-frequency* | Frequency of the first component in Hz. Range is 0 to 65535. |
| *2nd-frequency* | Frequency of the second component in Hz. Range is 0 to 65535. |
| *1st-amplitude-fxs* | (optional) Amplitude of the first component for FXS signaling in tenths of a decibel (dB). Range is -300 to 0. |
| *1st-amplitude-fxo* | (optional) Amplitude of the first component for FXO signaling in tenths of a decibel (dB). Range is -300 to 0. |
| *1st-amplitude-e&m* | (optional) Amplitude of the first component for E&M signaling in tenths of a decibel (dB). Range is -300 to 0. |
| *2nd-amplitude-fxs* | (optional) Amplitude of the second component for FXS signaling in tenths of a decibel (dB). Range is -300 to 0. |

| | |
|---|---|
| *2nd-amplitude-fxo* | (optional) Amplitude of the second component for FXO signaling in tenths of a decibel (dB). Range is -300 to 0. |
| *2nd-amplitude-e&m* | (optional) Amplitude of the second component for E&M signaling in tenths of a decibel (dB). Range is -300 to 0. |
| *1st-on* | On time for the first component in milliseconds (ms). Range is 0 to 65535. |
| *1st-off* | Off time for the first component in milliseconds (ms). Range is 0 to 65535. |
| *2nd-on* | On time for the second component in milliseconds (ms). Range is 0 to 65535. |
| *2nd-off* | Off time for the second component in milliseconds (ms). Range is 0 to 65535. |
| *3rd-on* | On time for the third component in milliseconds (ms). Range is 0 to 65535. |
| *3rd-off* | Off time for the third component in milliseconds (ms). Range is 0 to 65535. |
| *4th-on* | On time for the fourth component in milliseconds (ms). Range is 0 to 65535. |
| *4th-off* | Off time for the fourth component in milliseconds (ms). Range is 0 to 65535. |

### For DTMF Testing

| | |
|---|---|
| **dtmf** | Specifies DTMF tone testing. |
| *r0* | Frequency of the DTMF tone in row 0. Range is 0 to 65535. Typical DTMF setting is shown in Table 161. |
| *r1* | Frequency of the DTMF tone in row 1. Range is 0 to 65535. Typical DTMF setting is shown in Table 161. |
| *r2* | Frequency of the DTMF tone in row 2. Range is 0 to 65535. Typical DTMF setting is shown in Table 161. |
| *r3* | Frequency of the DTMF tone in row 3. Range is 0 to 65535. Typical DTMF setting is shown in Table 161. |
| *c0* | Frequency of the DTMF tone in column 0. Range is 0 to 65535. Typical DTMF setting is shown in Table 161. |
| *c1* | Frequency of the DTMF tone in column 1. Range is 0 to 65535. Typical DTMF setting is shown in Table 161. |
| *c2* | Frequency of the DTMF tone in column 2. Range is 0 to 65535. Typical DTMF setting is shown in Table 161. |
| *c3* | Frequency of the DTMF tone in column 3. Range is 0 to 65535. Typical DTMF setting is shown in Table 161. |
| *a1* | Frequency for the low frequency level. Range is 0 to 65535. |
| *a2* | Frequency for the high frequency level. Range is 0 to 65535. |

### For Pulse Ratio Testing

| | |
|---|---|
| **pulse** | Specifies pulse ratio testing. |
| *percent* | Percentage of the break period for a dialing pulse. Valid entries are numbers from 1 to 99. |

### To Show Test Details

| | |
|---|---|
| **show** | Displays the test voice tone parameters. |

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

| Release | Modification |
|---|---|
| 12.0(7)XK | This command was introduced. |

## Usage Guidelines

The two-letter country code used in the *locale* argument is based on the ISO-3166 standard. The codes used for this keyword are shown in Table 160.

***Table 160 Valid Command Entries for* locale *Argument***

| Country | *locale* Argument Value |
|---|---|
| Argentina | **ar** |
| Australia | **au** |
| Austria | **at** |
| Belgium | **be** |
| Brazil | **br** |
| Canada | **ca** |
| China | **cn** |
| Colombia | **co** |
| Czech Republic | **cz** |
| Denmark | **dk** |
| Egypt[1] | **eg** |
| Finland | **fi** |
| France | **fr** |
| Germany | **de** |
| Ghana[1] | **gh** |
| Great Britain | **gb** |
| Greece | **gr** |
| Hong Kong | **hk** |
| Hungary | **hu** |
| Iceland | **is** |
| India | **in** |
| Indonesia | **id** |
| Ireland | **ie** |
| Israel | **il** |
| Italy | **it** |

*Table 160 Valid Command Entries for* locale *Argument (continued)*

| Country | *locale* Argument Value |
|---|---|
| Japan | **jp** |
| Jordan[1] | **jo** |
| Kenya[1] | **ke** |
| Korea Republic | **kr** |
| Lebanon[1] | **lb** |
| Luxembourg | **lu** |
| Malaysia | **my** |
| Mexico | **mx** |
| Nepal[1] | **np** |
| Netherlands | **nl** |
| New Zealand | **nz** |
| Nigeria[1] | **ng** |
| Norway | **no** |
| Pakistan[1] | **pk** |
| Panama[1] | **pa** |
| Peru | **pe** |
| Philippines | **ph** |
| Poland | **pl** |
| Portugal | **pt** |
| Russian Federation | **ru** |
| Saudi Arabia[1] | **sa** |
| Singapore | **sg** |
| Slovakia | **sk** |
| Slovenia | **si** |
| South Africa | **za** |
| Spain | **es** |
| Sweden | **se** |
| Switzerland | **ch** |
| Taiwan | **tw** |
| Thailand | **th** |
| Turkey | **tr** |
| United States | **us** |
| Venezuela | **ve** |
| Zimbabwe[1] | **zw** |

1. Not applicable to Cisco MC3810 multiservice access concentrators with a prior Cisco IOS Release of 12.0(4)T.

Table 161 shows typical DTMF frequencies and the keypad entries to which they correspond.

***Table 161 Typical DTMF Frequencies***

| | Column ID | c0 | c1 | c2 | c3 |
|---|---|---|---|---|---|
| **Row ID** | **Frequency** | **1209** | **1336** | **1477** | **1633** |
| **r0** | **697** | 1 | 2 | 3 | A |
| **r1** | **770** | 4 | 5 | 6 | B |
| **r2** | **852** | 7 | 8 | 9 | C |
| **r3** | **941** | * | 0 | # | D |

## Examples

The following example shows the output for the **test voice tone us show** command. This command shows the current settings for the tones for a specified locale.

```
Router# test voice tone us show
Code:US   Country:United States
DTMF freq.(Hz) Row / col: 697,  770,  852,  941 / 1209, 1336, 1477, 1633
Pulse dial:normal, Percent make:40%, DTMF low Amp. = 65446,  high Amp. = 65467,   Pcm:u-Law
     Tone       NF   FOF  FOS  AOF_FXS AOF_FXO AOF_EM AOS_FXS AOS_FXO AOS_EM  ONTF  OFTF  ONTS  OFTS  ONTT  OFTT  ONT4  OFT4
BUSY            2   480   620    -170    -170    -240    -170    -170    -240   500   500     0     0     0     0     0     0
RING_BACK        2   440   480    -160    -160    -190    -160    -160    -190  2000  4000     0     0     0     0     0     0
CONGESTION       2   480   620    -170    -170    -190    -170    -170    -240   250   250     0     0     0     0     0     0
NUM_UNOBTAINAB  2   480   620    -170    -170    -190    -170    -170    -240   250   250     0     0     0     0     0     0
DIALTONE        2   350   440    -165    -165    -185    -165    -165    -185 65535     0     0     0     0     0     0     0
DIAL_TONE2       2  350   440    -165    -165    -185    -165    -165    -185 65535     0     0     0     0     0     0     0
OUT_OF_SERVICE 1  950    0    -150    -150    -185      0       0       0    330   330     0     0     0     0     0     0
ADDR_ACK        1  600    0    -240    -240    -240      0       0       0    125   125   125 65535     0     0     0     0
DISCONNECT      1  600    0    -150    -150    -185      0       0       0    330   330   330 65535     0     0     0     0
OFFHOOK_NOTICE  2  1400  2040    -240    -240    -240    -240    -240    -240   100   100     0     0     0     0     0     0
OFFHOOK_ALERT   2  1400  2040    -150    -150    -185    -150    -150    -185   100   100     0     0     0     0     0     0
```

## Related Commands

| Command | Description |
|---|---|
| **cptone** | Specifies a regional analog tone, ring, and cadence setting |

# test voice translation-rule

To test the functionality of a translation rule, use the **test voice translation-rule** command in privileged EXEC mode.

**test voice translation-rule** *number input-test-string* [**type** *match-type* [**plan** *match-type*] ]

## Syntax Description

| | |
|---|---|
| *number* | Specifies the number of the translation rule to be tested. Range is from 1 to 2147483647. |
| *input-test-string* | String to be tested by the translation rule. |
| **type** *match-type* | (Optional) Number type of the call. Valid values for the *match-type* argument are as follows:<br>• **abbreviated**—Abbreviated representation of the complete number as supported by this network.<br>• **any**—Any type of called number.<br>• **international**—Number called to reach a subscriber in another country.<br>• **national**—Number called to reach a subscriber in the same country, but outside the local network.<br>• **network**—Administrative or service number specific to the serving network.<br>• **reserved**—Reserved for extension.<br>• **subscriber**—Number called to reach a subscriber in the same local network.<br>• **unknown**—Number of a type that is unknown to the network. |
| **plan** *match-type* | (Optional) Numbering plan of the call. Valid values for the *match-type* argument are as follows:<br>• **any**—Any type of called number.<br>• **data**—Number called for data calls.<br>• **ermes**—European Radio Message standard numbering plan.<br>• **isdn**—Called number for an ISDN network.<br>• **national**—Number called to reach a subscriber in the same country, but outside the local network.<br>• **private**—Number called for a private network.<br>• **reserved**—Reserved for extension.<br>• **telex**—Numbering plan for Telex equipment.<br>• **unknown**—Number of a type that is unknown to the network. |

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

The number type and calling plan are optional parameters defined in a translation rule. If either parameter is defined, the call must match the match pattern and the type or plan value in order to be selected for translation.

## Examples

The following example tests the functionality of translation rule 5 with string 2015550101.

```
Router(config)# voice translation-rule 5

Router(cfg-translation-rule)# rule 1 /201/ /102/
Router(cfg-translation-rule)# exit

Router# test voice translation-rule 5 2015550101

Matched with rule 5
Original number:2015550101  Translated number:1025550101
Original number type: none    Translated number type: none
Original number plan: none    Translated number plan: none
```

The following examples display the error messages for a nonexistent rule or match pattern:

```
Router# test voice translation-rule 6 2015550101

Error: Ruleset 6 not found

Router# test voice translation-rule 5 2125550101

2125550101 Didn't match with any rules
```

## Related Commands

| Command | Description |
|---|---|
| **rule (voice translation-rule)** | Defines the translation-rule criteria. |
| **show voice translation-rule** | Displays the configuration for voice translation rules. |
| **voice translation-rule** | Initiates the translation-rule definition. |

# test vrm busyout

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

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

## Syntax Description

| | |
|---|---|
| *slot-number* | Slot in which the voice feature card (VFC) is installed. Range is from 0 to 11. There is no default value. |
| *first-dsp-number* | First DSP in a range to be busied out. Each VFC holds 96 DSPs, so the range is from 1 to 96. There is no default value. |
| *last-dsp-number* | Last DSP in a range to be busied out. Each VFC holds 96 DSPs, so the range is from 1 to 96. There is no default value. |
| **channel** | A certain channel on the specified DSPs is to be busied out. |
| *number* | Channel to be busied out. Values are 1 or 2. There is no default value. |
| **all** | All 96 DSPs on the VFC installed in the defined slot are to be busied out. |

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

| Release | Modification |
|---|---|
| 12.0(7)T | This command was introduced on Cisco AS5800. |

## Usage Guidelines

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

## Examples

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

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

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

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

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

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

## Related Commands

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

# test vrm reset

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

**test vrm reset** *slot-number dsp-number*

## Syntax Description

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

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

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

## Examples

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

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

# test vrm unbusyout

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

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

## Syntax Description

| | |
|---|---|
| *slot-number* | Number that identifies the slot in which the voice feature card (VFC) is installed. Range is from 0 to 11. There is no default value. |
| *first-dsp-number* | First DSP in a range to be restored. Each VFC holds 96 DSPs. Range is from 1 to 96. There is no default value. |
| *last-dsp-number* | Last DSP in a range to be restored. Each VFC holds 96 DSPs. Range is 1 to 96. There is no default value. |
| **channel** | A certain channel on the specified DSPs is to be restored. |
| *number* | Channel to be restored. Values are 1 or 2. There is no default value. |
| **all** | All 96 DSPs on the VFC installed in the defined slot are to be restored. |

## Defaults

No default behavior or values

## Command Modes

Privileged EXEC

## Command History

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

## Usage Guidelines

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

## Examples

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

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

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

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

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

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

## Related Commands

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

# tgrep address-family

To set the address family to be used on a local dial peer, use the **tgrep address-family** command in dial peer configuration mode. To return to the global setting, use the **no** form of this command.

**tgrep address family** {**e164** | **decimal** | **penta-decimal**}

**no tgrep address family** {**e164** | **decimal** | **penta-decimal**}

## Syntax Description

| | |
|---|---|
| **e164** | E.164 address family. |
| **decimal** | Decimal address family |
| **penta-decimal** | Penta-decimal address family |

## Defaults

No default behavior or values.

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(1) | This command was introduced. |

## Usage Guidelines

The E. 164 address family is used if the telephony network is a public telephony network. Decimal and pentadecimal options can be used to advertise private dial plans. For example if a company wants to use TRIP in within their enterprise telephony network using 5-digit extensions, then the gateway would advertise the beginning digits of their private numbers as a decimal address family. These calls cannot be sent out of the company's private telephony network because they are not E.164-compliant.

The pentadecimal family allows numbers 0 through 9 and alphabetic characters A through E and can be used in countries where letters are also carried in the called number.

## Examples

The following example shows that POTS dial peer 10 has the address family set for E.164 addresses:

```
Router(config)# dial-peer voice pots 10
Router(config-dial-peer)# tgrep address family e164
```

## Related Commands

| Command | Description |
|---|---|
| **dial-peer voice** | Enters dial-peer configuration mode and specifies the method of voice-related encapsulation. |

# tgrep advertise (dial peer)

To set the attributes for advertisement of the prefix on this dial peer or to disable advertisement on this dial peer altogether, use the **tgrep advertise** command in dial peer configuration mode. To return to using the global setting, use the **no** form of this command.

**tgrep advertise** [**csr**] [**ac**] [**tc**] [**carrier** | **trunk-group**] [**disable**]

**no tgrep advertise** [**csr**] [**ac**] [**tc**] [**carrier** | **trunk-group**] [**disable**]

## Syntax Description

| | |
|---|---|
| **csr** | Call success rate |
| **ac** | Available circuits |
| **tc** | Total circuits |
| **carrier** | Carrier code address family |
| **trunk-group** | Trunk group address family |
| **disable** | Disables advertisement of this dial peer |

## Defaults

Prefix advertisement is not sent.

## Command Modes

Dial peer configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(1) | This command was introduced. |

## Usage Guidelines

When only **tgrep advertise** is entered, the dial peer is advertised without any other attribute.

When **no tgrep advertise** is used on the dial peer, the dial peer inherits the attributes set in the global **advertise** command.

When the global **no advertise** command is used, it forbids advertisement of that particular address family altogether. The **tgrep advertise** command has no effect until the advertisement of the address family is enabled globally.

## Examples

The following example shows a TGREP advertisement that sends call success rate, available circuits, total circuits, and carrier address family attribute information:

```
Router(config)# dial-peer voice pots 10
Router(config-dial-peer)# tgrep advertise csr ac tc carrier
```

## Related Commands

| Command | Description |
|---|---|
| **dial-peer voice** | Enters dial-peer configuration mode and specifies the method of voice-related encapsulation. |

# tgrep advertise (trunk group)

To turn on the advertisement of this trunk group for resource availability and other carrier information, use the **tgrep advertise** command in trunk group configuration mode. To turn off local trunk group advertisement and use the global setting, use the **no** form of this command.

**tgrep advertise [csr] [ac] [tc] [disable]**

**no tgrep advertise [csr] [ac] [tc] [disable]**

## Syntax Description

| | |
|---|---|
| **csr** | Call success rate. |
| **ac** | Available circuits. |
| **tc** | Total circuits. |
| **disable** | Disables advertisement on the trunk group. |

## Defaults

Trunk group advertisement is not sent

## Command Modes

Trunk group configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(1) | This command was introduced. |

## Usage Guidelines

When only **tgrep advertise** is entered, the trunk group is advertised without any other attribute. When **no tgrep advertise** is used, the trunk group uses the global setting configured with the **advertise** command in TGREP configuration mode. To turn off advertisement of this trunk group, the **disable** keyword should be used.

There is a subtle difference between the **no** form of this command and the **no** form of the global **advertise** command. When **no tgrep advertise** is used on the trunk group, the trunk group inherits the attributes set in the global **advertise** command.

When the global **no advertise** command is used, it forbids advertisement of that particular address family altogether. The **tgrep advertise** command has no effect until the advertisement of the address family is enabled globally.

When the **carrier** keyword is used, the carrier defined under the trunk group assumes the configuration. Because multiple trunk groups can have the same carrier defined, the same configuration will show up under all trunk groups that have the same carrier defined. When the **no tgrep advertise carrier** command is used to revert to the global carrier configuration for the carrier under this trunk group, the same will happen to all the trunk groups who have the same carrier defined under them.

**Note** This command overrides the attributes set for advertisement using the global **advertise (tgrep)** command.

## Examples

The following example shows that trunk group 101 has been configured to send a TGREP advertisement that sends call success rate, available circuits, total circuits, and prefix attribute information:

```
Router(config)# trunk group 101
Router(config-dial-peer)# tgrep advertise csr ac tc carrier
```

## Related Commands

| Command | Description |
| --- | --- |
| **trunk group** | Defines the trunk group and enters trunk group configuration mode. |

# tgrep local-itad

To enable TGREP on the gateway and enter TGREP configuration mode, use the **tgrep local-itad** command in global configuration mode. To disable TRIP on the gateway, use the **no** form of this command.

**tgrep local-itad** *itad_number*

**no tgrep local-itad** *itad_number*

## Syntax Description

| | |
|---|---|
| *itad_number* | ITAD number associated with the gateway. The value can be from 1 to 4294967295. |

## Defaults

TGREP is not enabled on the gateway.

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.3(1) | This command was introduced. |

## Examples

The following example shows TGREP being enabled for ITAD number 1234:

```
Router(config)# tgrep local-itad 1234
```

## Related Commands

| Command | Description |
|---|---|
| **address-family** | Sets the global address family to be used on all dial peers. |
| **advertise (tgrep)** | Turns on reporting for a specified address family. |
| **neighbor** | Creates a TGREP session with another device. |

# threshold noise

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

**threshold noise** {*value*}

**no threshold noise** {*value*}

## Syntax Description

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

## Defaults

-62 dB

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(13b) | This command was introduced on the following platforms: Cisco 1700 Cisco 1751, Cisco 2600 (with and without the NM-HDA), Cisco 3600 (with and without the NM-HDA), Cisco 7200 (with and without the NM-HDA), Cisco AS5300, Cisco AS5800, and Cisco MC3810. |
| 12.2(16) | This command was integrated into Cisco IOS Release 12.2(16). |

## Usage Guidelines

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

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

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

## Examples

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

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

# time-format (cm-fallback)

To set the time-display format on all Cisco IP phones attached to a router, use the **time-format** command in call-manager-fallback configuration mode. To disable the time-display format, use the **no** form of this command.

**time-format** {**12** | **24**}

**no time-format** {**12** | **24**}

## Syntax Description

| | |
|---|---|
| **12** | 12-hour format. |
| **24** | 24-hour format. |

## Defaults

12-hour format

## Command Modes

Call-manager-fallback configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XT | This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers. |
| 12.2(8)T1 | This command was implemented on Cisco 2600-XM and Cisco 2691 routers. |
| 12.2(11)T | This command was implemented on the Cisco 1760. |

## Usage Guidelines

The **time-format** command sets the time display format on all the Cisco IP phones attached to the router.

## Examples

The following example shows the time format on the Cisco IP phones being set to the 24-hour format:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# time-format 24
```

## Related Commands

| Command | Description |
|---|---|
| **call-manager-fallback** | Enables SRS Telephony feature support and enters call-manager-fallback configuration mode. |

# timeout leg3

To set the timeout value for a leg 3 AAA preauthentication request, use the **timeout leg3** command in AAA preauthentication configuration mode. To return the timeout value to its default, use the **no** form of this command.

**timeout leg3** *milliseconds*

**no timeout leg3** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | Timeout value for leg 3 preauthentication, in milliseconds. Range is from 100 to 1000. The default is 100. |

## Defaults

100 milliseconds.

## Command Modes

AAA preauthentication configuration

## Command History

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

## Usage Guidelines

If the timeout timer expires before AAA has responded to a preauthentication request, the call is rejected.

The term *leg 3* refers to a call segment from the IP network to a terminating (outgoing) gateway that takes traffic from an IP network to a PSTN network.

## Examples

The following example sets the timeout for a leg 3 AAA preauthentication request to 250 milliseconds:

```
Router(config)# aaa preauth
Router(config-preauth)# timeout leg3 250
```

## Related Commands

| Command | Description |
|---|---|
| **aaa preauth** | Enters AAA preauthentication configuration mode. |

# timeout tcrit

To configure the critical timeout value, T(critical), for the interdigit timer used in digit map matching, use the **timeout tcrit** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tcrit** *tcrit-value*

**no timeout tcrit**

## Syntax Description

| | |
|---|---|
| *tcrit-value* | Critical timeout value, T(critical), in seconds. Range is from 1 to 600. Default is 4. |

## Defaults

4 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The interdigit timer is used when matching a digit map, which is a representation of the number and type of digits that a gateway can expect to collect in a buffer, based on the network dial plan. The interdigit timer is started when the first digit is entered and is restarted after each new digit is entered, until a digit map match or mismatch occurs.

The interdigit timer takes on one of two values, T(partial) or T(critical). When at least one more digit is required to make a match to any of the patterns in the digit map, the value of T(partial) is used for the timer. If a timer is all that is required to produce a match according to the digit map, T(critical) is used for the timer.

When the interdigit timer is used without a digit map, it takes on the value T(critical). It is started immediately and is simply canceled (but not restarted) as soon as a digit is entered.

## Examples

The following example sets the T(critical) value to 15 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tcrit 15
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |
| **timeout tpar** | Configures the MGCP partial timeout value, T(partial), for the interdigit timer used in digit map matching. |

# timeout tdinit

To configure the initial waiting delay value (Tdinit) for the disconnected procedure, use the **timeout tdinit** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tdinit** *tdinit-value*

**no timeout tdinit**

## Syntax Description

| | |
|---|---|
| *tdinit-value* | Initial waiting delay (Tdinit) for the disconnected procedure, in seconds. The disconnected timer is initialized to a randomly selected value between 0 and Tdinit. Range is from 1 to 30. Default is 15. |

## Defaults

15 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

When a gateway recognizes that an endpoint has lost its communication with the call agent (has become *disconnected)*, a timer known as the disconnected timer is initialized to a random value between 0 and the disconnected initial waiting delay (Tdinit), which is configured with the **timeout tdinit** command. The gateway then waits for one of three things: the end of this timer, the reception of a command from the call agent, or the detection of local user activity for the endpoint, such as an off-hook transition. When one of the first two cases occurs, the gateway initiates the *disconnected procedure* for that endpoint. In the third case, the detection of local user activity, a minimum waiting delay (Tdmin) also must have elapsed. This value is configured with the **timeout tdmin** command.

The disconnected procedure consists of the endpoint sending a RestartInProgress (RSIP) message to the call agent, stating that it was disconnected and is now trying to reestablish connectivity.

If the disconnected procedure is unsuccessful and the endpoint is still disconnected, the disconnected timer is doubled; this process is repeated until the timer value reaches the maximum waiting delay (Tdmax), which is configured with the **timeout tdmax** command.

## Examples

The following example sets the initial waiting delay value (Tdinit) to 25 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tdinit 25
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |
| **timeout tdmax** | Configures the maximum timeout for the MGCP disconnected procedure. |
| **timeout tdmin** | Configures the minimum timeout for the MGCP disconnected procedure. |

# timeout tdmax

To configure the maximum timeout value (Tdmax) for the disconnected procedure, use the **timeout tdmax** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tdmax** *tdmax-value*

**no timeout tdmax**

## Syntax Description

| | |
|---|---|
| *tdmax-value* | Maximum timeout value (Tdmax) for the disconnected procedure, in seconds. Range is from 300 to 600. The default is 600. |

## Defaults

600 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

When a gateway recognizes that an endpoint has lost its communication with the call agent (has become *disconnected),* a timer known as the disconnected timer is initialized to a random value between 0 and the disconnected initial waiting delay (Tdinit), which is configured with the **timeout tdinit** command. The gateway then waits for one of three things: the end of this timer, the reception of a command from the call agent, or the detection of local user activity for the endpoint, such as an off-hook transition. When one of the first two cases occurs, the gateway initiates the *disconnected procedure* for that endpoint. In the third case, the detection of local user activity, a minimum waiting delay (Tdmin) also must have elapsed. This value is configured with the **timeout tdmin** command.

The disconnected procedure consists of the endpoint sending a RestartInProgress (RSIP) message to the call agent, stating that it was disconnected and is now trying to reestablish connectivity.

If the disconnected procedure is unsuccessful and the endpoint is still disconnected, the disconnected timer is doubled; this process is repeated until the timer value reaches the maximum waiting delay (Tdmax), which is configured with the **timeout tdmax** command.

## Examples

The following example sets the maximum timeout value (Tdmax) to 450 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tdmax 450
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |
| **timeout tdinit** | Configures the initial timeout for the MGCP disconnected procedure. |
| **timeout tdmin** | Configures the minimum timeout for the MGCP disconnected procedure. |

# timeout tdmin

To configure the minimum timeout value (Tdmin) for the disconnected procedure, use the **timeout tdmin** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tdmin** *tdmin-value*

**no timeout tdmin**

## Syntax Description

| | |
|---|---|
| *tdmin-value* | Minimum timeout (Tdmin) for the disconnected procedure, in seconds. Range is from 1 to 30. The default is 15. |

## Defaults

15 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

When a gateway recognizes that an endpoint has lost its communication with the call agent (has become *disconnected),* a timer known as the disconnected timer is initialized to a random value between 0 and the disconnected initial waiting delay (Tdinit), which is configured with the **timeout tdinit** command. The gateway then waits for one of three things: the end of this timer, the reception of a command from the call agent, or the detection of local user activity for the endpoint, such as an off-hook transition. When one of the first two cases occurs, the gateway initiates the *disconnected procedure* for that endpoint. In the third case, the detection of local user activity, a minimum waiting delay (Tdmin) also must have elapsed. This value is configured with the **timeout tdmin** command.

The disconnected procedure consists of the endpoint sending a RestartInProgress (RSIP) message to the call agent, stating that it was disconnected and is now trying to reestablish connectivity.

If the disconnected procedure is unsuccessful and the endpoint is still disconnected, the disconnected timer is doubled; this process is repeated until the timer value reaches the maximum waiting delay (Tdmax), which is configured with the **timeout tdmax** command.

## Examples

The following example sets the minimum timeout value (Tdmin) to 20 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tdmin 20
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |
| **timeout tdinit** | Configures the initial timeout for the MGCP disconnected procedure. |
| **timeout tdmax** | Configures the maximum timeout for the MGCP disconnected procedure. |

# timeout thist

To configure the packet storage timeout value (Thist), use the **timeout thist** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout thist** *thist-value*

**no timeout thist**

## Syntax Description

| | |
|---|---|
| *thist-value* | Package storage timeout (Thist), in seconds. Range is from 1 to 60. The default is 30. |

## Defaults

30 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

MGCP messages are carried over User Datagram Protocol (UDP), and are therefore subject to packet loss. When a response to a message is not received promptly, the sender retransmits the message. The gateway keeps in memory a list of the responses it has sent for the number of seconds in the Thist timeout value. The gateway also keeps a list of the messages currently being processed, with their transaction identifiers, to prevent processing or acknowledging the same message more than once.

## Examples

The following example sets the packet storage timeout value (Thist) to 15 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout thist 15
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints, or to configure the default profile. |

# timeout tone busy

To configure the busy-tone timeout value, use the **timeout tone busy** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone busy** *busy-tone-value*

**no timeout tone busy**

## Syntax Description

| | |
|---|---|
| *busy-tone-value* | Busy-tone timeout, in seconds. Range is from 1 to 600. The default is 30. |

## Defaults

30 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the busy-tone timeout value when the call agent does not provide a timeout value associated with the request to generate a busy tone signal.

## Examples

The following example sets the busy tone timeout value to 45 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone busy 45
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tone cot1

To configure the continuity1 (cot1) tone timeout value, use the **timeout tone cot1** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone cot1** *cot1-tone-value*

**no timeout tone cot1**

## Syntax Description

| | |
|---|---|
| *cot1-tone-value* | Continuity1 (cot1) tone timeout, in seconds. Range is from 1 to 600. The default is 3. |

## Defaults

3 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the continuity1 (cot1) tone timeout value when the call agent does not provide a timeout value associated with the request to generate a cot1 tone signal.

Continuity1 and continuity2 tone signals are used in Integrated Services Digital Networks User Part (ISUP) calls to determine that a call path has been established before connecting a call. The call agent is provisioned to know which test to apply to a given endpoint.

## Examples

The following example sets the continuity1 tone timeout value to 25 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone cot1 25
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |
| **timeout tone cot2** | Sets the continuity2 tone timeout value for MGCP. |

# timeout tone cot2

To configure the continuity2 (cot2) tone timeout value, use the **timeout tone cot2** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone cot2** *cot2-tone-value*

**no timeout tone cot2**

## Syntax Description

| | |
|---|---|
| *cot2-tone-value* | Continuity2 (cot2) tone timeout, in seconds. Range is from 1 to 600. The default is 3. |

## Defaults

3 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the continuity2 (cot2) tone timeout value when the call agent does not provide a timeout value associated with the request to generate a cot2 tone signal.

Continuity1 and continuity2 tone signals are used in Integrated Services Digital Networks User Part (ISUP) calls to determine that a call path has been established before connecting a call. The call agent is provisioned to know which test to apply to a given endpoint.

## Examples

The following example sets the continuity2 tone timeout value to 50 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone cot2 50
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |
| **timeout tone cot1** | Sets the continuity1 tone timeout value for MGCP. |

# timeout tone dial

To configure the dial tone timeout value, use the **timeout tone dial** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone dial** *dial-tone-value*

**no timeout tone dial**

## Syntax Description

| | |
|---|---|
| *dial-tone-value* | Dial tone timeout value, in seconds. Range is from 1 to 600. The default is 16. |

## Defaults

16 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the dial tone timeout value when the call agent does not provide a timeout value associated with the request to generate a dial tone signal.

## Examples

The following example sets the dial tone timeout value to 25 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone dial 25
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tone dial stutter

To configure the stutter dial tone timeout value, use the **timeout tone dial stutter** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone dial stutter** *stutter-value*

**no timeout tone dial stutter**

## Syntax Description

| | |
|---|---|
| *stutter-value* | Timeout value for the stutter dial tone, in seconds. Range is from1 to 600. The default is 16. |

## Defaults

16 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the stutter dial tone timeout value when the call agent does not provide a timeout value associated with the request to generate a stutter dial tone signal.

## Examples

The following example sets the stutter dial tone timeout value to 25 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone dial stutter 25
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tone mwi

To configure the timeout value for the message-waiting indicator tone, use the **timeout tone mwi** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone mwi** *mwi-tone-value*

**no timeout tone mwi**

## Syntax Description

| | |
|---|---|
| *mwi-tone-value* | Message-waiting-indicator (MWI) tone timeout value, in seconds. Range is from 1 to 600. The default is 16. |

## Defaults

16 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the *mwi-tone-value* when the call agent does not provide a timeout value for a request to generate the message-waiting indicator tone signal.

## Examples

The following example sets the timeout value for the message-waiting indicator tone to 100 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone mwi 100
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tone network congestion

To configure the network congestion tone timeout value, use the **timeout tone network congestion** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone network congestion** *congestion-tone-value*

**no timeout tone network congestion**

## Syntax Description

| | |
|---|---|
| *congestion-tone-value* | Network-congestion tone timeout value, in seconds. Range is from 1 to 600. The default is 180. |

## Defaults

180 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the congestion tone timeout value when the call agent does not provide a timeout value associated with the request to generate a network congestion tone signal.

## Examples

The following example sets the network congestion tone timeout value to 240 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone network congestion 240
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tone reorder

To configure the reorder tone timeout value, use the **timeout tone reorder** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone reorder** *reorder-tone-value*

**no timeout tone reorder**

## Syntax Description

| | |
|---|---|
| *reorder-tone-value* | Reorder-tone timeout value, in seconds. Range is from 1 to 600. The default is 30. |

## Defaults

30 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the reorder tone timeout value when the call agent does not provide a timeout value associated with the request to generate a reorder tone signal.

## Examples

The following example sets the reorder tone timeout value to 60 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone reorder 60
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tone ringback

To configure the ringback tone timeout value, use the **timeout tone ringback** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone ringback** *ringback-tone-value*

**no timeout tone ringback**

## Syntax Description

| | |
|---|---|
| *ringback-tone-value* | Ringback-tone timeout value, in seconds. Range is from 1 to 600. The default is 180. |

## Defaults

180 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the ringback tone timeout value when the call agent does not provide a timeout value associated with the request to generate a ringback tone signal.

## Examples

The following example sets the ringback tone timeout value to 120 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone ringback 120
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tone ringback connection

To configure the timeout value for the ringback tone on connection, use the **timeout tone ringback connection** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone ringback connection** *connect-tone-value*

**no timeout tone ringback connection**

## Syntax Description

| | |
|---|---|
| *connect-tone-value* | Timeout value for the ringback tone on connection, in seconds. Range is from 1 to 600. The default is 180. |

## Defaults

180 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses this value when the call agent does not provide a timeout value associated with the request to generate the ringback tone signal on connection.

## Examples

The following example sets the timeout value for the ringback tone on connection to 120 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone ringback connection 120
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tone ringing

To configure the ringing tone timeout value, use the **timeout tone ringing** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone ringing** *ringing-tone-value*

**no timeout tone ringing**

## Syntax Description

| | |
|---|---|
| *ringing-tone-value* | Ringing tone timeout value, in seconds. Range is from 1 to 600. The default is 180. |

## Defaults

180 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the ringing tone timeout value when the call agent does not provide a timeout value associated with the request to generate a ringing tone signal.

## Examples

The following example sets the ringing tone timeout value to 240 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone ringing 240
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tone ringing distinctive

To configure the distinctive ringing tone timeout value, use the **timeout tone ringing distinctive** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tone ringing distinctive** *distinct-tone-value*

**no timeout tone ringing distinctive**

## Syntax Description

| | |
|---|---|
| *distinct-tone-value* | Distinctive-ringing tone timeout value, in seconds. Range is from 1 to 600. the default is 180. |

## Defaults

180 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the distinctive ringing tone timeout value when the call agent does not provide a timeout value associated with the request to generate a signal for distinctive ringing.

## Examples

The following example sets the distinctive ringing tone timeout value to 240 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tone ringing distinctive 240
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeout tpar

To configure the partial timeout value, T(partial), for the interdigit timer used in digit map matching, use the **timeout tpar** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tpar** *tpar-value*

**no timeout tpar**

## Syntax Description

| | |
|---|---|
| *tpar-value* | Partial timeout value, T(partial), in seconds. Range is from 1 to 60. The default is 16. |

## Defaults

16 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The interdigit timer is used when matching digit maps. It is started when the first digit is entered, and is restarted after each new digit is entered, until a digit map match or mismatch occurs.

The interdigit timer takes on one of two values, T(partial) or T(critical). When at least one more digit is required to make a match to any of the patterns in the digit map, the value of T(partial) is used for the timer. If a timer is all that is required to produce a match according to the digit map, T(critical) is used for the timer.

When the interdigit timer is used without a digit map, it takes on the value T(critical). It is started immediately and is simply canceled (but not restarted) as soon as a digit is entered.

## Examples

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

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tpar 15
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |
| **timeout tcrit** | Configures the MGCP critical timeout value, T(critical), for the interdigit timer used in digit map matching. |

# timeout tsmax

To configure the maximum timeout value after which MGCP messages are removed from the retransmission queue, use the **timeout tsmax** command in MGCP profile configuration mode. To reset to the default, use the **no** form of this command.

**timeout tsmax** *tsmax-value*

**no timeout tsmax**

## Syntax Description

| | |
|---|---|
| *tsmax-value* | Timeout value for MGCP messages to be removed from the retransmission queue, in seconds. Range is from 1 to 100. The default is 20. |

## Defaults

20 seconds

## Command Modes

MGCP profile 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. |
| 12.2(11)T | This command was implemented on the Cisco AS5300 and Cisco AS5850. |

## Usage Guidelines

This command is used when configuring values for a Media Gateway Control Protocol (MGCP) profile.

The gateway uses the *tsmax-value* argument to determine how long to store MGCP messages before they are removed from the retransmission queue.

## Examples

The following example sets the timeout value for the maximum retransmission of MGCP messages to 45 seconds:

```
Router(config)# mgcp profile nyc-ca
Router(config-mgcp-profile)# timeout tsmax 45
```

## Related Commands

| Command | Description |
|---|---|
| **mgcp** | Starts and allocates resources for the MGCP daemon. |
| **mgcp profile** | Initiates MGCP profile mode to create and configure a named MGCP profile associated with one or more endpoints or to configure the default profile. |

# timeouts call-disconnect

To configure the delay time for which a Foreign Exchange Office (FXO) voice port waits before disconnecting an incoming call after disconnect tones are detected, use the **timeouts call-disconnect** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

**timeouts call-disconnect** {*seconds* | **infinity**}

**no timeouts call-disconnect**

## Syntax Description

| | |
|---|---|
| *seconds* | Duration in seconds for which an FXO voice port stays in the connected state after the voice port detects a disconnect tone. Range is 1 to 120. The default is 60. |
| **infinity** | Disables disconnect supervision. The voice port does not disconnect when a disconnect tone is detected. |

## Defaults

60 seconds

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(9)T | This command was introduced on Cisco 3600 series routers. |
| 12.0(4)T | This command was introduced on Cisco 3600 series routers. |
| 12.2(2)T | This command was implemented on Cisco 1750, Cisco 2600 series, and Cisco MC3810. The **infinity** keyword was added. |

## Usage Guidelines

Use this command to change the time for which an FXO voice port remains connected after the calling party hangs up, when a call is not answered. Use of the **infinity** keyword is not recommended for disabling the disconnect supervision feature.

## Examples

The following example configures voice port 1/1 on a Cisco MC3810 multiservice access concentrator to remain connected for 2 seconds while a disconnect tone is received by the voice port:

```
voice-port 1/1
 timeouts call-disconnect 2
```

The following example configures voice port 0/0/1 on a Cisco 3600 to remain connected for 3 seconds while a disconnect tone is received by the voice port:

```
voice-port 0/0/1
 timeouts call-disconnect 3
```

## 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** | Specifies the delay time for releasing the calling voice port after a disconnect tone is received from the called voice port. |
| **timing delay-duration** | Configures the delay dial signal duration for a specified voice port. |

# timeouts initial

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

**timeouts initial** *seconds*

**no timeouts initial** *seconds*

## Syntax Description

| | |
|---|---|
| *seconds* | Initial timeout duration, in seconds. Range is 0 to 120. The default is 10. |

## Defaults

10 seconds

## Command Modes

Voice-port configuration

## Command History

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

## Usage Guidelines

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

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

## Examples

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

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

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

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

## Related Commands

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

# timeouts interdigit (cm-fallback)

To configure the interdigit timeout value for all Cisco IP phones attached to a router, use the **timeouts interdigit** command in call-manager-fallback configuration mode. To disable the interdigit timeout value, use the **no** form of this command.

**timeouts interdigit** *seconds*

**no timeouts interdigit** *seconds*

## Syntax Description

| | |
|---|---|
| *seconds* | Interdigit timeout duration, in seconds, for all the Cisco IP phones. Range is 2 to 120. The default is 10. |

## Defaults

10 seconds

## Command Modes

Call-manager-fallback configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco IAD2420, and Cisco 7200. |
| 12.2(2)XT | This command was implemented on Cisco 1750 and Cisco 1751. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers. |
| 12.2(8)T1 | This command was implemented on Cisco 2600-XM and Cisco 2691 routers. |
| 12.2(11)T | This command was implemented on the Cisco 1760. |

## Usage Guidelines

This command specifies how long, in seconds, the system waits after a caller enters the initial digit or a subsequent digit of the dialed string. The timeouts interdigit timer is activated when the caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is terminated. The default is 10 seconds.

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

## Examples

The following example sets the interdigit timeout value to 5 seconds for all Cisco IP phones:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# timeouts interdigit 5
```

In this example, the 5 seconds refers to the elapsed time after which an incompletely dialed number times out. For example, if you dial nine digits (408555898) instead of the required 10 digits (4085558984), you hear a busy tone after 5 “timeout” seconds.

## Related Commands

| Command | Description |
| --- | --- |
| **call-manager-fallback** | Enables SRS Telephony feature support and enters call-manager-fallback configuration mode. |
| **timeouts interdigit** | Configures the interdigit timeout value for a specified voice port. |

# timeouts interdigit (telephony-service)

To configure the interdigit timeout value for all Cisco IP phones attached to a router, use the **timeouts interdigit** command in telephony-service configuration mode. To disable the interdigit timeout value, use the **no** form of this command.

**timeouts interdigit** *seconds*

**no timeouts interdigit** *seconds*

## Syntax Description

| | |
|---|---|
| *seconds* | Interdigit timeout duration, in seconds, set on the timer for all the Cisco IP phones. Range is from 2 to 120 seconds. The default is 10 seconds. |

## Defaults

10 seconds

## Command Modes

Telephony-service configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced on the Cisco 2600 series, Cisco 360 series, and Cisco IAD2420. |
| 12.2(2)XT | This command was implemented on Cisco 1750 and Cisco 1751. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 3725 and Cisco 3745 routers. |
| 12.2(8)T1 | This command was implemented on Cisco 2600-XM and Cisco 2691 routers. |
| 12.2(11)T | This command was implemented on the Cisco 1760. |

## Usage Guidelines

This command specifies how long, in seconds, the system waits after a caller enters the initial digit or a subsequent digit of the dialed string. The timeouts interdigit timer is activated when the caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is terminated. The default is 10 seconds.

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

## Examples

The following example sets the interdigit timeout value to 5 seconds for all Cisco IP phones:

```
Router(config)# telephony-service
Router(config-telephony-service)# timeouts interdigit 5
```

In this example, the 5 seconds refers to the elapsed time after which an incompletely dialed number times out. For example, if you dial nine digits (4085559898) instead of the required 10 digits (4085559898), you hear a busy tone after 5 “timeout” seconds.

## Related Commands

| Command | Description |
|---|---|
| **ephone-dn** | Enters ephone-dn configuration mode. |
| **telephony-service** | Enables Cisco IOS Telephony Service and enters telephony-service configuration mode. |
| **timeouts interdigit** | Configures the interdigit timeout value for a specified voice port. |

# timeouts interdigit (voice port)

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

**timeouts interdigit** *seconds*

**no timeouts interdigit** *seconds*

## Syntax Description

| | |
|---|---|
| *seconds* | Interdigit timeout duration, in seconds. Range is 0 to 120. The default is 10. |

## Defaults

10 seconds

## Command Modes

Voice-port configuration

## Command History

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

## Usage Guidelines

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

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

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

## Examples

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

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

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

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

## Related Commands

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

# timeouts ringing

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

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

**no timeouts ringing**

## Syntax Description

| | |
|---|---|
| *seconds* | Duration, in seconds, for which a voice port allows ringing to continue if a call is not answered. Range is 5 to 60000. The default is 180. |
| **infinity** | Ringing continues until the caller goes on-hook. |

## Defaults

180 seconds

## Command Modes

Voice-port configuration

## Command History

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

## Usage Guidelines

This command allows you to limit the length of time for which a caller can continue ringing a telephone when there is no answer.

## Examples

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

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

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

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

## Related Commands

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

# timeouts wait-release

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

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

**no timeouts wait-release**

## Syntax Description

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

## Defaults

30 seconds

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1) MA | This command was introduced on 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

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

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

## Examples

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

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

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

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

## Related Commands

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

# timer accessrequest sequential delay

To configure the intermessage delay used when a border element (BE) is trying to determine a route from a list of neighboring BEs, use the **timer accessrequest sequential delay** command in Annex G configuration mode. To reset the default value, use the **no** form of this command.

**timer accessrequest sequential delay** *value*

**no timer**

## Syntax Description

| | |
|---|---|
| *value* | Amount of allowed intermessage delay (in increments of 100 ms). Range is from 0 to 10. The default is 1 (100 ms). |

## Defaults

1 (100 ms)

## 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 the Cisco AS5850. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T. |

## Usage Guidelines

Setting the value of the delay to 0 causes the BE to broadcast or “blast” the AccessRequest messages to all eligible neighbors.

## Examples

The following example shows a timer delay of 1000 ms.

```
Router(config)# call-router h323-annexg be20
Router(config-annexg)# timer accessrequest sequential delay 10
```

## Related Commands

| Command | Description |
|---|---|
| **call-router** | Enables the Annex G border element configuration commands. |

# timer cluster-element announce

To define the time interval between successive announcement messages that are exchanged between elements of a local cluster, use the **timer cluster-element announce** command in gatekeeper configuration mode. To reset to the default, use the **no** form of this command.

**timer cluster-element announce** *seconds*

**no timer cluster-element announce**

## Syntax Description

| | |
|---|---|
| *seconds* | Number of seconds between announcement periods. When a gatekeeper comes on line, it announces its presence on a periodic basis. The default is 30 seconds. |

## Defaults

30 seconds

## Command Modes

Gatekeeper configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(5)XM | This command was introduced. |
| 12.2(2)T | This command was integrated into Cisco IOS Release 12.2(2)T. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |

## Usage Guidelines

The announcement indication is exchanged at a set interval of time and carries information about the call and endpoint capacity for the zone. This allows the alternate gatekeepers to manage the bandwidth for a single zone even though the gatekeepers are in separate physical devices.

The gatekeeper assumes that the alternate gatekeeper has failed (and assumes that any previously allocated bandwidth is now available) if the gatekeeper does not receive an announcement message within six announcement periods or if the TCP connection with the gatekeeper is detected to be broken.

Lower this interval for closer tracking between elements. Raise it to lower messaging overhead.

## Examples

The following command sets the announcement period to 20 seconds:

```
timer cluster-element announce 20
```

The following command resets the announcement period to the default value:

```
no timer cluster-element announce
```

## Related Commands

| Command | Description |
|---|---|
| **zone cluster local** | Defines a local grouping of gatekeepers. |
| **zone remote** | Statically specifies a remote zone if DNS is unavailable or undesirable. |

# timer irr period

To configure the information request response (IRR) timer, or the periodic interval of IRR messages sent by the gatekeeper, use the **timer irr period** command in gatekeeper configuration mode. To disable, use the **no** form of this command.

**timer irr period** *minutes*

**no timer irr period**

## Syntax Description

| | |
|---|---|
| *minutes* | Length, in minutes, of the interval between IRR messages. Range is from 1 to 60. The default is 4. |

## Defaults

4 minutes

## Command Modes

Gatekeeper configuration

## Command History

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

## Usage Guidelines

Use this command to configure IRR frequency that is included in the admission confirm (ACF) message. The IRR frequency is set to 240 seconds (4 minutes), based on an average 4-minute call hold time. The IRR allows the gatekeepers to terminate calls for which a disengage request (DRQ) has not been received. If missing DRQs are not a problem, the IRR frequency can be set to a larger value than 4 minutes, minimizing the number of unnecessary IRRs sent by a gateway.

## Examples

The following example shows that the IRR timer has been configured with a value of 45, meaning that IRR messages are sent by the gatekeeper every 45 minutes:

```
gatekeeper
.
.
.
lrq reject-resource-low
 no irq global-request
 timer lrq seq delay 10
 timer lrq window 6
 timer irr period 45
 no shutdown
```

## Related Commands

| Command | Description |
|---|---|
| **timer lrq seq delay** | Defines the time interval between successive LRQ messages. |
| **timer lrq window** | Defines the time window during which the gatekeeper collects responses to one or more outstanding LRQs. |
| **timer server timeout** | Specifies the timeout value for a response from a back-end GKTMP server. |

# timer lrq seq delay

To define the time interval between successive sequential location requests (LRQs), use the **timer lrq seq delay** command in gatekeeper configuration mode. To reset to the default, use the **no** form of this command.

**timer lrq seq delay** *time*

**no timer lrq seq delay**

## Syntax Description

| | |
|---|---|
| *time* | Time interval, in 100-millisecond units. Range is 1 to 10 (0.1 to 1 second). The default is 5 (500 milliseconds). |

## Defaults

5 units (500 milliseconds)

## Command Modes

Gatekeeper configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(5)XM | This command was introduced. |
| 12.2(2)T | This command was integrated into Cisco IOS Release 12.2(2)T. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |

## Usage Guidelines

The LRQ sequential timing source (SEQ) delay is used to set the time between sending LRQs to remote gatekeepers for address resolution. To resolve an address, the gatekeeper might have several remote zones configured, and it can send the LRQs simultaneously (blast) or sequentially (seq). The gatekeeper chooses the best route based on availability and cost. Using LRQs sequentially results in lower network traffic, but it can increase latency of calls when the most preferred route is unavailable.

Lowering the time increases traffic on the network but might reduce the call setup time.

## Examples

The following command sets the LRQ delay timer to 100 milliseconds:

```
timer lrq seq delay 1
```

The following command resets the LRQ delay timer to the default value:

```
no timer lrq seq delay
```

## Related Commands

| Command | Description |
|---|---|
| **timer lrq window** | Defines the time window during which the gatekeeper collects responses to one or more outstanding LRQs. |

# timer lrq window

To define the time window during which the gatekeeper collects responses to one or more outstanding LRQs, use the **timer lrq window** command in gatekeeper configuration mode. To reset to the default, use the **no** form of this command.

**timer lrq window** *seconds*

**no timer lrq window**

## Syntax Description

| | |
|---|---|
| *seconds* | Time window, in seconds. Range is 1 to 15. The default is 3. |

## Defaults

3 seconds

## Command Modes

Gatekeeper configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(5)XM | This command was introduced. |
| 12.2(2)T | This command was integrated into Cisco IOS Release 12.2(2)T. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |

## Usage Guidelines

Increasing the time can increase the call success rate but might reduce the overall time for call setup.

## Examples

The following command sets the timer to 5 seconds:

```
timer lrq window 5
```

The following command sets the timer to the default value:

```
no timer lrq window
```

## Related Commands

| Command | Description |
|---|---|
| **timer lrq seq delay** | Defines the time interval between successive sequential LRQs. |

# timer receive-rtcp

To enable the Real-Time Control Protocol (RTCP) timer and to configure a multiplication factor for the RTCP timer interval for Session Initiation Protocol (SIP) or H.323, use the **timer receive-rtcp** command in gateway configuration mode. To reset to the default, use the **no** form of this command.

**timer receive-rtcp** *timer*

**no timer receive-rtcp**

## Syntax Description

| | |
|---|---|
| *timer* | Multiples of the RTCP report transmission interval. Range is 2 to 1000. The default is 5. |

## Defaults

5 multiples

## Command Modes

Gateway configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release. |
| 12.2(11)T | This command was applicable to the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release. |

## Usage Guidelines

When the **ip rtcp report interval** and **timer receive-rtcp** commands are used, the gateway uses RTCP report detection, rather than Real-Time Protocol (RTP) packet detection, to determine whether calls on the gateway are still active or should be disconnected. RTCP report detection is therefore more reliable than RTP packet detection because there can be periods during voice calls when one or both parties are not sending RTP packets.

One common example of a voice session in which no RTP is sent is when a caller dials into a conference call and mutes his endpoint. If voice activity detection (VAD, also known as silence suppression) is enabled, no RTP packets are sent while the endpoint is muted. However, the muted endpoint continues to send RTCP reports at the interval specified by the **ip rtcp report interval** command.

The **timer receive-rtcp** *timer* argument (or Mfactor for multiplication factor) is multiplied by the interval that is set using the **ip rtcp report interval** command. If no RTCP packets are received during the calculated interval, the call is disconnected. The gateway signals the disconnect to the VoIP network and the TDM network so that upstream and downstream devices can clear their resources. The gateway sends a Q.931 DISCONNECT to the TDM network and a SIP BYE or H.323 ReleaseComplete to the VoIP network to clear the call when the timer expires. The Q.931 DISCONNECT is sent with a cause code value of 3 (no route) for SIP calls and a cause code value of 41 (temporary failure) for H.323 calls. No Q.931 Progress Indicator (PI) value is included in the DISCONNECT.

To show timer-related output for SIP calls, use the **debug ccsip events** command. To show timer-related output for H.323 calls, use the **debug cch323 h225** command.

## Examples

The following example sets the multiplication factor to 10 (or *x* * 10, where *x* is the interval that is set with the **ip rtcp report interval** command):

```
Router(config)# gateway
Router(config-gateway)# timer receive-rtcp 10
Router(config-gateway)# exit
```

## Related Commands

| Command | Description |
|---|---|
| **debug cch323 h225** | Traces the state transition of the gateway H.225 state machine based on the processed events. |
| **debug ccsip events** | Displays all SIP SPI events tracing and traces the events posted to SIP SPI from all interfaces. |
| **ip rtcp report interval** | Configures the minimum interval of RTCP report transmissions. |

# timer server retry

To set the gatekeeper's retry timer for failed Gatekeeper Transaction Message Protocol (GKTMP) connections, use the **timer server retry** command in gatekeeper configuration mode. To reset the timer to its default, use the **no** form of this command or the **default server timer retry** command.

**server timer retry** *seconds*

**no server timer retry**

**default server timer retry**

## Syntax Description

| | |
|---|---|
| *seconds* | Number of seconds for which the gatekeeper should wait before retrying the GKTMP server. Range is from 1 through 300. The default is 30. |

## Defaults

30 seconds

## Command Modes

Gatekeeper configuration

## Command History

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

## Usage Guidelines

After the gatekeeper detects that its GKTMP server TCP connection has failed, the gatekeeper retries the server after an interval based on the setting of this timer, and keeps retrying until the connection is established.

This timer applies only to deployments where static triggers are used between the gatekeeper and the GKTMP server. If dynamic triggers are used, the server must determine and implement a retry mechanism if the TCP connection to the gatekeeper fails.

## Examples

The following example shows that the retry timer has been set to 45 seconds:

```
Router# show gatekeeper configuration
.
.
.
h323id tet
 gw-type-prefix 1#* default-technology
 gw-type-prefix 9#* gw ipaddr 1.1.1.1 1720
 timer server retry 45
 no shutdown
.
.
.
```

## Related Commands

| Command | Description |
|---|---|
| **timer server timeout** | Specifies the timeout value for a response from a back-end GKTMP server. |

# timer server timeout

To specify the timeout interval for a response from a back-end Gatekeeper Transaction Message Protocol (GKTMP) application server, use the **timer server timeout** command in gatekeeper configuration mode. To reset to the default, use the **no** form of this command.

**timer server timeout** *time*

**no timer server timeout**

## Syntax Description

| | |
|---|---|
| *time* | Timeout interval, in 100-millisecond units. Range is 1 to 50 (0.1 to 5 seconds). The default is 3 (300 milliseconds). |

## Defaults

3 units (300 milliseconds)

## Command Modes

Gatekeeper configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(2)XM | This command was introduced. |
| 12.2(2)T | This command was integrated into Cisco IOS Release 12.2(2)T. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |

## Examples

The following command sets the timeout interval to 400 milliseconds:

```
timer server timeout 4
```

The following command resets the timeout interval to the default value:

```
no timer server timeout
```

## Related Commands

| Command | Description |
|---|---|
| **server registration-port** | Configures the listener port for the server to establish a connection with the gatekeeper. |
| **server trigger** | Configures a static server trigger for external applications. |

# timers

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

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

**no timers**

## Syntax Description

| | |
|---|---|
| **trying** *number* | Time (in milliseconds) to wait for a 100 response to an INVITE request. Range is from 100 to 1000. The default is 500. |
| **connect** *number* | Time (in milliseconds) to wait for a 200 response to an ACK request. Range is from 100 to 1000. The default is 500. |
| **disconnect** *number* | Time (in milliseconds) to wait for a 200 response to a BYE request. Range is from 100 to 1000. The default is 500. |
| **expires** *number* | Time (in milliseconds) for which an INVITE request is valid. Range is from 60000 to 300000. The default is 180000. |

## Defaults

**trying**, **connect**, and **disconnect**—500 ms
**expires**—180000 ms

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced. |
| 12.1(3)T | This command was modified to change the names of the parameters. Two of the parameters (**invite-wait-180** and **invite-wait-200**) were combined into one (**trying**). |
| 12.2(2)XA | This command was implemented on the Cisco AS5400 and AS5350. |
| 12.2(2)XB1 | This command was implemented on the Cisco AS5850. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 7200 series routers. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release. |
| 12.2(11)T | This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release. |

## Usage Guidelines

If you used an earlier version of this command to configure timers, the timer settings are maintained. The output of the **show running-config** command reflects both previous and current timers.

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

## Examples

The following example sets the trying timers to the default of 500 ms.

```
Router(config)# sip-ua
Router(config-sip-ua)# timers trying 500
```

## Related Commands

| Command | Description |
|---|---|
| **default** | Sets a command to its default. |
| **inband-alerting** | Specifies an inband-alerting SIP header. |
| **max-forwards** | Specifies the maximum number of hops for a request. |
| **retry (SIP user-agent)** | Configures the SIP signaling timers for retry attempts. |
| **transport** | Enables SIP UA transport for TCP/UDP. |

# timers comet

To set how long the Session Initiation Protocol (SIP) user agent (UA) waits before retransmitting conditions-met (COMET) requests, use the **timers comet** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

**timers comet** *time*

**no timers comet**

## Syntax Description

| | |
|---|---|
| *time* | Waiting time, in milliseconds. Range is 100 to 1000. The default is 500. |

## Defaults

500 milliseconds

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release. |
| 12.2(11)T | This command was applicable to the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release. |

## Usage Guidelines

COMET, or conditions met, indicates whether preconditions for a given call or session have been met. This command is applicable only with calls involving quality of service (QoS) (calls other than best-effort).

## Examples

The following example sets retransmission time to 500 milliseconds:

```
Router(config)# sip-ua
Router(config-sip-ua)# timers comet 500
```

## Related Commands

| Command | Description |
|---|---|
| **show sip-ua statistics** | Displays response, traffic, timer, and retry statistics. |
| **show sip-ua timers** | Displays the current settings for SIP UA timers. |
| **timers prack** | Sets how long the UA waits before retransmitting a PRACK request. |

# timers connect

To set how long the Session Initiation Protocol (SIP) user agent (UA) waits for a 200 response to an ACK request, use the **timers connect** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

**timers connect** *number*

**no timers connect** *number*

## Syntax Description

| | |
|---|---|
| *number* | Waiting time, in milliseconds. Range is from 100 to 1000. The default is 500. |

## Defaults

500 milliseconds

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced on Cisco 2600 series, Cisco 3600 series, and Cisco AS5300. |
| 12.1(3)T | This command was modified to change the names of the parameters. Two of the parameters (invite-wait-180 and invite-wait-200) were combined into one (trying). |
| 12.2(2)XA | This command was implemented on the Cisco AS5350 and Cisco AS5400. |
| 12.2(2)XB1 | This command was implemented on the Cisco AS5850. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release. |

## Usage Guidelines

If you used the previous more generic **timers** command to configure timers, your previous timer settings are maintained. The output of the **show running-config** command reflects both timers.

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

## Examples

The following example sets connect time to 200 milliseconds:

```
sip-ua
 timers connect 200
```

## Related Commands

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

# timers disconnect

To set how long the Session Initiation Protocol (SIP) user agent (UA) waits for a 200 response to a BYE request, use the **timers** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

**timers disconnect** *time*

**no timers disconnect** *time*

## Syntax Description

| | |
|---|---|
| *time* | Waiting time, in milliseconds. Range is 100 to 1000. The default is 500. |

## Defaults

500 milliseconds

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced on Cisco 2600 series, Cisco 3600 series, and Cisco AS5300. |
| 12.1(3)T | This command was modified to change the names of the parameters. Two of the parameters (invite-wait-180 and invite-wait-200) were combined into one (trying). |
| 12.2(2)XA | This command was implemented on the Cisco AS5350 and Cisco AS400. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 7200 series. Supported for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms is not included in this release. |

## Usage Guidelines

If you used the previous more generic **timers** command to configure timers, your previous timer settings are maintained. The output of the **show running-config** command reflects both timers.

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

## Examples

The following example sets disconnect time to 200 milliseconds:

```
sip-ua
 timers disconnect 200
```

## Related Commands

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

# timers expires

To set how long a Session Initiation Protocol (SIP) INVITE request is valid, use the **timers** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

**timers expires** *time*

**no timers expires**

## Syntax Description

| | |
|---|---|
| *time* | Expiration time, in milliseconds. Range is 60000 to 300000. The default is 180000. |

## Defaults

180,000 milliseconds

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced on Cisco 2600 series, Cisco 3600 series, and Cisco AS5300. |
| 12.1(3)T | This command was modified to change the names of the parameters. Two of the parameters (invite-wait-180 and invite-wait-200) were combined into one (trying). |
| 12.2(2)XA | This command was implemented on Cisco AS5350 and Cisco AS5400. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |
| 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 is not included in this release. |

## Usage Guidelines

If you used the previous more generic **timers** command to configure timers, your previous timer settings are maintained. The output of the **show running-config** command reflects both timers.

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

## Examples

The following example sets the expiration time to 180,000 milliseconds:

```
sip-ua
 timers expires 180000
```

## Related Commands

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

# timers hold

To enable the Session Initiation Protocol (SIP) hold timer and configure the timer interval before disconnecting a held call, use the **timers hold** command in SIP user-agent configuration mode. To restore the default value, use the **no** form of this command.

**timers hold** *time*

**no timers hold**

## Syntax Description

| | |
|---|---|
| *time* | Specifies the time (in minutes) to wait before sending a BYE request. Range is from 15 to 2880 minutes. The default is 2880. |

## Defaults

Enabled

*time*: 2880 minutes

## Command Modes

SIP user-agent configuration mode

## Command History

| Release | Modification |
|---|---|
| 12.3(1) | This command was introduced. |

## Usage Guidelines

The hold timer is typically activated when a gateway receives a call hold request from the other endpoint, for example, a SIP phone.

## Examples

The following example sets the hold timer to expire after 75 minutes:

```
Router(config-sip-ua)# timers hold 75
```

## Related Commands

| Command | Description |
|---|---|
| **show sip-ua timers** | Displays the current settings for SIP user agent timers. |
| **suspend-resume** | Enables SIP Suspend and Resume (call-hold) functionality. |
| **timer receive-rtcp** | Enables media inactivity Real-Time Control Protocol (RTCP) timer. |

# timers notify

To set how long the Session Initiation Protocol (SIP) user agent (UA) waits before retransmitting a Notify message, use the **timers notify** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

**timers notify** *time*

**no timers notify**

## Syntax Description

| | |
|---|---|
| *time* | Waiting time, in milliseconds. Range is 100 to 1000. The default is 500. |

## Defaults

500 milliseconds

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(2)XB2 | This command was implemented on Cisco AS5850. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release. |

## Usage Guidelines

A Notify message informs the user agent that initiated the transfer or Refer request about the outcome of the SIP transaction.

## Examples

The following example sets retransmission time to 500 milliseconds:

```
Router(config)# sip-ua
Router(config-sip-ua)# timers notify 500
```

## Related Commands

| Command | Description |
|---|---|
| **show sip-ua statistics** | Displays response, traffic, timer, and retry statistics |
| **show sip-ua timers** | Displays the current settings for SIP UA timers |

# timers prack

To set how long the Session Initiation Protocol (SIP) user agent (UA) wait s before retransmitting a provisional response acknowledgement (PRACK) request, use the **timers prack** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

**timers prack** *time*

**no timers prack**

## Syntax Description

| | |
|---|---|
| *time* | Waiting time, in milliseconds. Range is 100 to 1000. The default is 500. |

## Defaults

500 milliseconds

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release. |
| 12.2(11)T | This command was applicable to the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release. |

## Usage Guidelines

PRACK allows reliable exchanges of SIP provisional responses between SIP endpoints. When the retransmission value is set, retransmissions are sent with an exponential backoff of up to 4 seconds. That is, the retransmission interval for each packet increases exponentially until 4 seconds is reached.

## Examples

The following example sets retransmission time to 500 milliseconds:

```
Router(config)# sip-ua
Router(config-sip-ua)# timers prack 500
```

## Related Commands

| Command | Description |
|---|---|
| **show sip-ua statistics** | Displays response, traffic, timer, and retry statistics. |
| **show sip-ua timers** | Displays the current settings for SIP UA timers. |
| **timers comet** | Sets how long the UA waits before retransmitting a COMET request. |

# timers refer

To set how long the Session Initiation Protocol (SIP) user agent (UA) waits before retransmitting a Refer request, use the **timers refer** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

**timers refer** *time*

**no timers refer**

## Syntax Description

| | |
|---|---|
| *time* | Waiting time, in milliseconds. Range is from 100 to 1000. Default is 500. |

## Defaults

500 milliseconds

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(11)YT | This command was introduced. |
| 12.2(15)T | This command is supported on the Cisco 1700 series, Cisco 2600 series, Cisco 3600 series, and the Cisco 7200 series routers in this release. |

## Usage Guidelines

A SIP Refer request is sent by the originating gateway to the receiving gateway and initiates call forward and call transfer capabilities.

## Examples

The following example sets retransmission time to 500 milliseconds:

```
Router(config)# sip-ua
Router(config-sip-ua)# timers refer 500
```

## Related Commands

| Command | Description |
|---|---|
| **show sip-ua statistics** | Displays response, traffic, timer, and retry statistics. |
| **show sip-ua timers** | Displays the current settings for SIP UA timers. |

# timers rel1xx

To set how long the Session Initiation Protocol (SIP) user agent (UA) waits before retransmitting a reliable1xx response, use the **timers rel1xx** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

**timers rel1xx** *time*

**no timers rel1xx**

## Syntax Description

| | |
|---|---|
| *time* | Waiting time, in milliseconds. Range is 100 to 1000. The default is 500. |

## Defaults

500 milliseconds

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release. |
| 12.2(11)T | This command was applicable to the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release. |

## Examples

The following example sets retransmission time to 400 milliseconds:

```
Router(config)# sip-ua
Router(config-sip-ua)# timers rel1xx 400
```

## Related Commands

| Command | Description |
|---|---|
| **retry rel1xx** | Configures how many times the reliable1xx response is retransmitted. |
| **show sip-ua statistics** | Displays response, traffic, timer, and retry statistics. |
| **show sip-ua timers** | Displays the current settings for SIP UA timers. |

# timers trying

To set how long the Session Initiation Protocol (SIP) user agent (UA) wait s for a 100 response to a SIP INVITE request, use the **timers** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

**timers trying** *time*

**no timers trying**

## Syntax Description

| | |
|---|---|
| *time* | Waiting time, in milliseconds. Range is 100 to 1000. The default is 500. |

## Defaults

500 milliseconds

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced on the Cisco 2600, Cisco 3600, and Cisco AS5300. |
| 12.1(3)T | This command was modified to change the names of the parameters. Two of the parameters (invite-wait-180 and invite-wait-200) were combined into one (trying). |
| 12.2(2)XA | This command was implemented on Cisco AS5350 and Cisco AS5400. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |
| 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 is not included in this release. |

## Usage Guidelines

If you used the previous more generic **timers** command to configure timers, your previous timer settings are maintained. The output of the **show running-config** command reflects both timers.

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

## Examples

The following example sets trying time to 500 milliseconds.

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

## Related Commands

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

# time-webedit (telephony-service)

To enable time setting through the web interface, use the **time-webedit** command in telephony-service configuration mode. To disable this feature, use the **no** form of this command.

**time-webedit**

**no time-webedit**

## Syntax Description

This command has no arguments or keywords.

## Defaults

No default behavior or values

## Command Modes

Telephony-service configuration

## Command History

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

## Usage Guidelines

The **time-webedit** command allows a local administrator of the Cisco IOS Telephony Service router to change and set time through the graphical user interface (GUI).

## Examples

The following example enables web editing of time:

```
Router(config)# telephony-service
Router(config-telephony-service)# time-webedit
```

## Related Commands

| Command | Description |
|---|---|
| **dn-webedit** | Enables adding of directory numbers through a web interface. |
| **telephony-service** | Enables Cisco IOS Telephony Service and enters telephony-service configuration mode. |

# timing clear-wait

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

**timing clear-wait** *time*

**no timing clear-wait** *time*

## Syntax Description

| | |
|---|---|
| *time* | Minimum time, in milliseconds, between an inactive seizure signal and the call being cleared. Cisco 3600 series range is from 200 to 2000. Cisco MC3810 range is from 100 to 2000. The default for both is 400. |

## Defaults

400 milliseconds

## Command Modes

Voice-port configuration

## Command History

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

## Usage Guidelines

This command is supported on E&M ports only.

## Examples

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

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

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

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

## Related Commands

| Command | Description |
|---|---|
| **timeouts initial** | Configures the initial digit timeout value for a specified voice port. |
| **timeouts interdigit** | Configures the interdigit timeout value for a specified voice port. |
| **timeouts wait-release** | Configures the timeout value for releasing voice ports on the Cisco MC3810. |
| **timing delay-duration** | Specifies the delay signal duration for a specified voice port. |

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

# timing delay-duration

To specify the delay signal duration for a specified voice port, use the **timing delay-duration** command in voice-port configuration mode. To reset 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 100 to 5000. The default is 2000. |

## Defaults

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 Cisco 3600 series voice port to 3000 milliseconds:

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

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

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

## Related Commands

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

| Command | Description |
|---|---|
| **timing delay-start** | Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port. |
| **timing delay-with-integrity** | Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810. |
| **timing dialout-delay** | Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810. |
| **timing dial-pulse min-delay** | Specifies the time between wink-like pulses for a specified voice port. |
| **timing digit** | Specifies the DTMF digit signal duration for a specified voice port. |
| **timing interdigit** | Specifies the DTMF interdigit duration for a specified voice port. |
| **timing percentbreak** | Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810. |
| **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 20 to 2000. The default on the Cisco 3600 series is 300. The default on the Cisco MC3810 is 150. |

## Defaults

Cisco 3600 series: 300 milliseconds
Cisco MC3810 multiservice concentrator: 150 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 Cisco 3600 series voice port to 250 milliseconds:

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

The following example sets the delay-start duration on a Cisco MC3810 voice port to 250 milliseconds:

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

## Related Commands

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

| Command | Description |
|---|---|
| **timing delay-duration** | Specifies the delay signal duration for a specified voice port. |
| **timing delay-with-integrity** | Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810. |
| **timing dialout-delay** | Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810. |
| **timing dial-pulse min-delay** | Specifies the time between wink-like pulses for a specified voice port. |
| **timing digit** | Specifies the DTMF digit signal duration for a specified voice port. |
| **timing interdigit** | Specifies the DTMF interdigit duration for a specified voice port. |
| **timing percentbreak** | Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810. |
| **timing pulse** | Specifies the pulse dialing rate for a specified voice port. |
| **timing pulse-interdigit** | Specifies the pulse interdigit timing for a specified voice port. |
| **timing wink-duration** | Specifies the maximum wink signal duration for a specified voice port. |
| **timing wink-wait** | Specifies the maximum wink-wait duration for a specified voice port. |

# timing delay-with-integrity

To specify the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810, use the **timing delay-with-integrity** command in voice-port configuration mode. To reset 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. |

## Defaults

0 milliseconds

## Command Modes

Voice-port configuration

## Command History

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

## Usage Guidelines

This command applies only to the Cisco MC3810. It is supported on E&M ports only.

## Examples

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

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

## Related Commands

| Command | Description |
|---|---|
| **timeouts initial** | Configures the initial digit timeout value for a specified voice port. |
| **timeouts interdigit** | Configures the interdigit timeout value for a specified voice port. |
| **timeouts wait-release** | Configures the timeout value for releasing voice ports on the Cisco MC3810. |
| **timing clear-wait** | Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port. |
| **timing delay-duration** | Specifies the delay signal duration for a specified voice port. |
| **timing delay-start** | Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port. |
| **timing dialout-delay** | Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810. |

| 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 on the Cisco MC3810. |
| **timing pulse** | Specifies the pulse dialing rate for a specified voice port. |
| **timing pulse-interdigit** | Specifies the pulse interdigit timing for a specified voice port. |
| **timing wink-duration** | Specifies the maximum wink signal duration for a specified voice port. |
| **timing wink-wait** | Specifies the maximum wink-wait duration for a specified voice port. |

# timing dialout-delay

To specify the dial-out delay for the sending digit on a specified voice port on the Cisco MC3810 multiservice concentrator, use the **timing dialout-delay** command in voice-port configuration mode. To reset 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. |

## Defaults

300 milliseconds

## Command Modes

Voice-port configuration

## Command History

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

## Usage Guidelines

This command applies only to the Cisco MC3810.

## Examples

The following example sets the dial-out delay on a Cisco MC3810 voice port to 350 milliseconds:

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

## Related Commands

| Command | Description |
|---|---|
| **timeouts initial** | Configures the initial digit timeout value for a specified voice port. |
| **timeouts interdigit** | Configures the interdigit timeout value for a specified voice port. |
| **timeouts wait-release** | Configures the timeout value for releasing voice ports on the Cisco MC3810. |
| **timing clear-wait** | Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port. |
| **timing delay-duration** | Specifies the delay signal duration for a specified voice port. |
| **timing delay-start** | Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port. |
| **timing delay-with-integrity** | Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810. |

| 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 on the Cisco MC3810. |
| **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 0 to 5000. The default is 300. |

## Defaults

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 Cisco 3600 series 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 on the Cisco MC3810. |
| **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 on the Cisco MC3810. |
| **timing dialout-delay** | Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810. |
| **timing digit** | Specifies the DTMF digit signal duration for a specified voice port. |
| **timing interdigit** | Specifies the DTMF interdigit duration for a specified voice port. |
| **timing percentbreak** | Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810. |
| **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 0 to 100. The default is 100. |

## Defaults

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 Cisco 3600 series voice port to 50 milliseconds:

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

The following example sets the DTMF digit signal duration on a Cisco MC3810 voice port to 50 milliseconds:

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

## Related Commands

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

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

# timing guard-out

To specify the guard-out duration of an Foreign Exchange Office (FXO) voice port, use the **timing guard-out** command in voice-port configuration mode. To 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. Range is from 300 to 3000. The default is 2000. |

## Defaults

2000 milliseconds

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)MA | This command was introduced on 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 applies to the Cisco 2600 and Cisco 3600 series routers and Cisco MC3810. This command is supported on FXO voice ports only.

## Examples

The following example sets the timing guard-out duration on a Cisco MC3810 voice port to 1000 milliseconds:

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

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

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

# timing hookflash-input

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

**timing hookflash-input** *milliseconds*

**no timing hookflash-input**

## Syntax Description

| | |
|---|---|
| *milliseconds* | Duration of the hookflash, in milliseconds. Range is 50 to 1550 milliseconds. Default is 600 milliseconds. |

## Defaults

600 milliseconds

## Command Modes

Voice-port configuration

## Command History

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

## Usage Guidelines

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

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

## Examples

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

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

## Related Commands

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

# timing hookflash-output

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

**timing hookflash-output** *time*

**no timing hookflash-output**

## Syntax Description

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

## Defaults

400 milliseconds

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced on Cisco 2500, Cisco 2600 series, Cisco 3600 series, Cisco 7200, 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-output** 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-output 200
```

**Related Commands**

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

# timing interdigit

To specify the dual-tone multifrequency (DTMF) interdigit duration for a specified voice port, use the **timing interdigit** command in voice-port configuration mode. To reset 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 50 to 500. The default is 100. |

## Defaults

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 Cisco 3600 series voice port to 150 milliseconds:

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

The following example sets the DTMF interdigit duration on a Cisco MC3810 voice port to 150 milliseconds:

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

## Related Commands

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

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

# timing percentbreak

To specify the percentage of the break period for dialing pulses for a voice port, use the **timing percentbreak** command in voice-port configuration mode. To reset 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. |

## Defaults

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 Cisco MC3810 voice port to 30 percent:

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

The following example sets the break period percentage on a Cisco 2600 series or Cisco 3600 series 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 10 to 20. The default is 20. |

## Defaults

20 pulses per seconds

## Command Modes

Voice-port configuration

## Command History

| Release | Modification |
|---|---|
| 11.3(1)T | This command was introduced on Cisco 3600 series. |
| 11.3(1)MA | This command was supported on 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 Cisco 3600 series voice port to 15 pulses per second:

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

The following example sets the pulse dialing rate on a Cisco MC3810 voice port to 15 pulses per second:

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

## Related Commands

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

| Command | Description |
|---|---|
| **timing delay-start** | Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port. |
| **timing delay-with-integrity** | Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810. |
| **timing dialout-delay** | Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810. |
| **timing dial-pulse min-delay** | Specifies the time between wink-like pulses for a specified voice port. |
| **timing digit** | Specifies the DTMF digit signal duration for a specified voice port. |
| **timing interdigit** | Specifies the DTMF interdigit duration for a specified voice port. |
| **timing percentbreak** | Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810. |
| **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 100 to 1000. The default is 500. |

## Defaults

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 Cisco 3600 series voice port to 300 milliseconds:

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

The following example sets the pulse-dialing interdigit timing on a Cisco MC3810 voice port to 300 milliseconds:

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

## Related Commands

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

| Command | Description |
|---|---|
| **timing delay-duration** | Specifies the delay signal duration for a specified voice port. |
| **timing delay-start** | Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port. |
| **timing delay-with-integrity** | Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810. |
| **timing dialout-delay** | Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810. |
| **timing dial-pulse min-delay** | Specifies the time between wink-like pulses for a specified voice port. |
| **timing digit** | Specifies the DTMF digit signal duration for a specified voice port. |
| **timing interdigit** | Specifies the DTMF interdigit duration for a specified voice port. |
| **timing percentbreak** | Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810. |
| **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 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 100 to 5000 milliseconds (ms). Supported on ear and mouth (E&M) ports only. |

## Defaults

*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. |

## Examples

The following example configures the maximum time to wait for wink signaling after an outgoing seizure is sent on a Cisco 3600 series voice port for 300 milliseconds:

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

The following example configures the maximum time to wait for wink signaling after an outgoing seizure is sent on a Cisco MC3810 multiservice concentrator voice port for 300 milliseconds:

```
voice-port 1/1
 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 on the Cisco MC3810 multiservice concentrator. |
| **timing clear-wait** | Indicates the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port. |
| **timing delay-duration** | Specifies the delay signal duration for a specified voice port. |

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

# timing wink-duration

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

**timing wink-duration** *time*

**no timing wink-duration** *time*

## Syntax Description

| | |
|---|---|
| *time* | Maximum wink-signal duration, in milliseconds, for a wink-start signal. Range is from 100 to 400. The default is 200. |

## Defaults

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. |

## 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.

## Examples

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

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

The following example sets the wink-signal duration on a Cisco MC3810 voice port to 300 milliseconds:

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

## Related Commands

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

| Command | Description |
|---|---|
| **timing delay-start** | Specifies the minimum delay time from outgoing seizure to out-dial address for a specified voice port. |
| **timing delay-with-integrity** | Specifies the duration of the wink pulse for the delay dial for a specified voice port on the Cisco MC3810. |
| **timing dialout-delay** | Specifies the dialout delay for the sending digit on a specified voice port on the Cisco MC3810. |
| **timing delay-with-integrity** | Specifies the time between wink-like pulses for a specified voice port. |
| **timing digit** | Specifies the DTMF digit signal duration for a specified voice port. |
| **timing interdigit** | Specifies the DTMF interdigit duration for a specified voice port. |
| **timing percentbreak** | Specifies the percentage of a break period for a dialing pulse for a specified voice port on the Cisco MC3810. |
| **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 100 to 5000. The default is 200. |

## Defaults

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. |

## 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 Cisco 3600 series voice port to 300 milliseconds:

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

The following example sets the wink-wait duration on a Cisco MC3810 voice port to 300 milliseconds:

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

## Related Commands

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

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

# token-root-name

To specify which root or Certificate Authority (CA) certificate the router uses to validate the settlement token in the incoming setup message, use the **token-root-name** command in settlement configuration mode. 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. |

## Defaults

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

## Command Modes

Settlement configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced on Cisco 2600 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. |
| **crypto ca trusted-root** | Configures the root certificate that the server uses to sign the settlement tokens. |
| **show settlement** | Displays the configuration for all settlement server transactions. |

# tone ringback alert-no-PI

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

**tone ringback alert-no-PI**

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

## Syntax Description

This command has no arguments or keywords.

## Defaults

No default behavior or values

## Command Modes

Dial-peer configuration

## Command History

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

## Usage Guidelines

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

## Examples

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

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

## Related Commands

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

# transfer-mode

To specify the type of call transfer for an individual IP phone directory number using the ITU-T H.450.2 standard, use the **transfer-mode** command in ephone-dn configuration mode. To remove this specification, use the **no** form of this command.

**transfer-mode** {**blind** | **consult**}

**no transfer-mode**

## Syntax Description

| | |
|---|---|
| **blind** | Transfers calls without consultation using a single phone line. |
| **consult** | Transfers calls with consultation using a second phone line, if available. |

## Defaults

No default behavior or values

## Command Modes

Ephone-dn configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(11)YT | This command was introduced. |
| 12.2(15)T | This command was integrated into Cisco IOS Release 12.2(15)T. |

## Usage Guidelines

This command specifies the type of call transfer for an individual Cisco IP phone line that is using the ITU-T H.450.2 protocol. It allows you to override the system default **transfer-system** setting (full-consult or full-blind) for that line.

Call transfers using H.450.2 can be blind or consultative. A blind transfer is one in which the transferring phone connects the caller to a destination line before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

You can specify blind or consultative transfer on a systemwide basis with the **transfer-system** command. The systemwide setting can then be overridden for individual phone lines with the **transfer-mode** command. For example, in an ITS network that is set up for consultative transfer, a specific line with an auto-attendant that automatically transfers incoming calls to specific extension numbers can be set to use blind transfer, because auto-attendants do not use consultative transfer.

Use this command with Cisco IOS Telephony Service (ITS) V2.1 or a later version.

## Examples

The following example sets blind mode for call transfers from this directory number:

```
Router(config)# ephone-dn 21354
Router(config-ephone-dn)# transfer-mode blind
```

## Related Commands

| Command | Description |
|---|---|
| **ephone-dn** | Enters ephone-dn configuration mode to set directory numbers and parameters for individual Cisco IP phone lines. |
| **transfer-system** | Specifies the call transfer method for all IP phones on a Cisco ITS router using the ITU-T H.450.2 standard. |

# transfer-pattern (cm-fallback)

To allow transfer of telephone calls from Cisco IP phones to phones other than Cisco IP phones, use the **transfer-pattern** command in call-manager-fallback configuration mode. To disable these transfers, use the **no** form of this command.

**transfer-pattern** *transfer-pattern*

**no transfer-pattern**

## Syntax Description

| | |
|---|---|
| *transfer-pattern* | String of digits for permitted call transfers. Wildcards are allowed. |

## Defaults

Transfer of calls is enabled only to Cisco IP phones.

## Command Modes

Call-manager-fallback configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(5)YD | This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420. |
| 12.2(2)XT | This command was implemented on Cisco 1750 and Cisco 1751. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers. |
| 12.2(8)T1 | This command was implemented on Cisco 2600-XM and Cisco 2691 routers. |
| 12.2(11)T | This command was implemented on the Cisco 1760. |

## Usage Guidelines

This command allows you to transfer calls to “other” phones—that is, to non-IP phones and phones outside of your network. A call is then established between the transferred party and the new recipient. By default, all Cisco IP phone directory numbers or virtual voice ports are allowed as transfer targets.

## Examples

The following example sets a transfer pattern:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# transfer-pattern 55540..
```

A maximum of 32 transfer patterns can be entered. In this example, 55540.. (the two periods are wildcards) permits transfers to any number in the range 555-4000 to 555-4099.

## Related Commands

| Command | Description |
|---|---|
| **call-manager-fallback** | Enables SRS Telephony feature support and enters call-manager-fallback configuration mode. |

# transfer-pattern (telephony-service)

To allow transfer of telephone calls from Cisco IP phones to phones other than Cisco IP phones, use the **transfer-pattern** command in telephony-service configuration mode. To disable these transfers, use the **no** form of this command.

**transfer-pattern** *transfer-pattern* **[blind]**

**no transfer-pattern**

## Syntax Description

| | |
|---|---|
| *transfer-pattern* | String of digits for permitted call transfers. Wildcards are allowed. A maximum of 32 transfer patterns can be entered, using a separate command for each one. |
| **blind** | When using H.450.2 consultative call transfer, forces transfers that match the pattern to be executed as blind transfers. Overrides settings made with the **transfer-system** and **transfer-mode** commands. |

## Defaults

Transer of calls is enabled only to local Cisco IP phones.

## Command Modes

Telephony-service configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(5)YD | This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420. |
| 12.2(2)XT | This command was implemented on Cisco 1750 and Cisco 1751. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 3725 and Cisco 3745 routers. |
| 12.2(8)T1 | This command was implemented on Cisco 2600-XM and Cisco 2691 routers. |
| 12.2(11)T | This command was implemented on the Cisco 1760. |
| 12.2(15)T | The **blind** keyword was added. |

## Usage Guidelines

This command allows you to transfer calls to “other” phones—that is, to non-IP phones and phones outside of your network. A call is then established between the transferred party and the new recipient. By default, all Cisco IP phone directory numbers are allowed as transfer targets.

The **blind** keyword is valid only for systems using Cisco IOS Telephony Service V2.1 or a later version and only applies to consultative transfers made using the H.450.2 standard. The **blind** keyword forces calls that are transferred to number matching the transfer pattern to be executed as blind or full-blind transfers, overriding any settings made with the **transfer-system** and **transfer-mode** commands.

## Examples

The following example sets a transfer pattern:

```
Router(config)# telephony-service
Router(config-telephony-service)# transfer-pattern 55540..
```

A maximum of 32 transfer patterns can be entered. In this example, 55540.. (the two periods are wildcards) permits transfers to any number in the range 555-4000 to 555-4099.

## Related Commands

| Command | Description |
|---|---|
| **ephone** | Enters ephone configuration mode. |
| **ephone-dn** | Enters ephone-dn configuration mode. |
| **telephony-service** | Enables Cisco IOS Telephony Service and enters telephony-service configuration mode. |

# transfer-system

To specify the call transfer method for all IP phones on a Cisco IOS Telephony Service (ITS) router using the ITU-T H.450.2 standard, use the **transfer-system** command in telephony-service configuration mode. To disable the call transfer method, use the **no** form of this command.

**transfer-system** {**blind** | **full-blind** | **full-consult** | **local-consult**}

**no transfer-system**

## Syntax Description

| | |
|---|---|
| **blind** | Transfers calls without consultation using a single phone line and the Cisco proprietary method. |
| **full-blind** | Transfers calls without consultation using H.450.2 standard methods. |
| **full-consult** | Transfers calls using H.450.2 with consultation using the second phone line if available, or the calls fall back to **full-blind** if the second line is unavailable. |
| **local-consult** | Transfers calls with local consultation using the second phone line if available, or the calls fall back to **blind** for nonlocal consultation or transfer target. This mode is intended for use primarily in Voice over Frame Relay (VoFR) networks, because the Cisco VoFR call transfer protocol does not support an end-to-end transfer with consultation mechanism. |

## Defaults

**blind**

## Command Modes

Telephony-service configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(11)YT | This command was introduced. |
| 12.2(15)T | This command was integrated into Cisco IOS Release 12.2(15)T. |

## Usage Guidelines

Use this command with Cisco ITS V2.1 or a later version.

Call transfers using the H.450.2 standard can be blind or consultative. A blind transfer is one in which the transferring phone connects the caller to a destination line before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party. When H.450.2 call transfer is selected using the **full-blind** or **full-consult** keyword, the router must be configured with a Tool Command Language (TCL) script that supports the H.450.3 protocol. The TCL script is loaded on the ITS router with the **call application voice** command.

You can specify blind or consultative transfer on a systemwide basis with the **transfer-system** command. The systemwide setting can then be overridden for individual phone lines with the **transfer-mode** command. For example, in an ITS network that is set up for consultative transfer, a specific line with an auto-attendant that automatically transfers incoming calls to specific extension numbers can be set to use blind transfer, because auto-attendants cannot use consultative transfer.

## Examples

The following example sets full consultation as the call transfer method for this ITS phone network:

```
Router(config)# telephony-service
Router(config-telephony-service)# transfer-system full-consult
```

## Related Commands

| Command | Description |
|---|---|
| **call application voice** | Defines an application, indicates the location of the corresponding TCL files that implement the application, and loads the selected TCL script. |
| **telephony-service** | Enables ITS and enters telephony-service configuration mode. |
| **transfer-mode** | Specifies the type of call transfer for an individual IP phone directory number using the H.450.2 standard. |

# 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. |

## Defaults

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 (cm-fallback)

To apply a translation rule to manipulate digits dialed by users of Cisco IP phones while CallManager fallback is active, use the **translate** command in call-manager-fallback configuration mode. To disable the translation rule, use the **no** form of this command.

**translate** {**called** | **calling**} *translation-rule-tag*

**no translate** {**called** | **calling**} *translation-rule-tag*

## Syntax Description

| | |
|---|---|
| **called** | Translation rule to apply to the number called by a Cisco IP phone. |
| **calling** | Translation rule to apply to the calling party number sent in the call setup message for calls originated from a Cisco IP phone. |
| *translation-rule-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. |

## Defaults

No default behavior or values

## Command Modes

Call-manager-fallback configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XT | This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers. |
| 12.2(8)T1 | This command was implemented on Cisco 2600-XM and Cisco 2691 routers. |
| 12.2(11)T | This command was implemented on the Cisco 1760. |

## Usage Guidelines

This command allows you to apply a preconfigured digit-translation rule to modify the number dialed by a specific extension (ephone-dn). 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 sets translation rule 20 to the inbound called number:

```
Router(config)# translation-rule 20
Router(config-translate)# rule 0 1234 2345 abbreviated
Router(config)# call-manager-fallback
Router(config-cm-fallback) translate called 20
```

## Related Commands

| Command | Description |
|---|---|
| **call-manager-fallback** | Enables Survivable Remote Site (SRS) Telephony feature support and enters call-manager-fallback configuration mode. |
| **translation-rule** | Creates a translation name and enters translation-rule configuration mode. |

# translate (ephone-dn)

To apply a translation rule to manipulate digits dialed by users of Cisco IP phones, use the **translate** command in ephone-dn configuration mode. To disable the translation rule, use the **no** form of this command.

**translate** {**called** | **calling**} *translation-rule-tag*

**no translate** {**called** | **calling**} *translation-rule-tag*

## Syntax Description

| | |
|---|---|
| **called** | Translate the called number. |
| **calling** | Translate the calling number. |
| *translation-rule-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. |

## Defaults

No default behavior or values

## Command Modes

Ephone-dn configuration

## Command History

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

## Usage Guidelines

This command allows you to select a preconfigured translation rule to modify the number dialed by a specific extension (Cisco IP phone destination number, or ephone-dn). 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. The translation rules are applied to the voice ports created by the ephone-dn. The **called** keyword translates the called number, and the **calling** keyword translates the calling number.

**Note** For this command to take effect, appropriate translation rules should have been created at the VoIP configuration level.

## Examples

The following example sets translation rule 20 to a number called from a Cisco IP phone:

```
Router(config)# translation-rule 20
Router(config-translate)# rule 0 1234 2345 abbreviated
Router(config)# ephone-dn 1
Router(config-ephone-dn) translate called 20
```

## Related Commands

| Command | Description |
|---|---|
| **ephone-dn** | Enters ephone-dn configuration mode. |
| **translation-rule** | Creates a translation name and enters translation-rule configuration mode. |

# 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**} *translation-rule-number*

**no translate** {**called** | **calling** | **redirect-called**} *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. |
| *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. |

## Defaults

No default behavior or values

## Command Modes

Voice translation-profile configuration

## 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. |

## Usage Guidelines

Use the **translate** 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. |

## Defaults

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. |

## Defaults

No default behavior or values

## Command Modes

Dial-peer 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 dial peer.

## Examples

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

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

## 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. |

## Defaults

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. |

## Defaults

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. |

## Defaults

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. |

## Defaults

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. |
| **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. |

## Defaults

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:<br>• VoIP Cisco 2600, Cisco 3600, and Cisco MC3810<br>• VoFR Cisco 2600, Cisco 3600, and Cisco MC3810<br>• VoATM Cisco 3600 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, Cisco 3600, Cisco AS5300, Cisco 7200, and Cisco 7500) |
| 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:<br>• VoIP Cisco MC3810<br>• VoFR Cisco 2600, Cisco 3600, and Cisco MC3810<br>• VoATM Cisco 3600 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

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

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

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

## Syntax Description

| | |
|---|---|
| **udp** | SIP user agent receives SIP messages on UDP port 5060. |
| **tcp** | SIP user agent receives SIP messages on TCP port 5060. |

## Defaults

Both UDP and TCP transport protocols are enabled.

## Command Modes

SIP user-agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.1(1)T | This command was introduced on Cisco 2600 series, Cisco 3600 series, and Cisco AS5300. |
| 12.1(3)T | This command was integrated into Cisco IOS release 12.1(3)T. |
| 12.2(2)XA | This command was implemented on Cisco AS5400 and Cisco AS5350. |
| 12.2(2)XB1 | This command was implemented on Cisco AS5850. |
| 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 is not included in this release. |
| 12.2(11)T | This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release. |

## Usage Guidelines

This command controls whether messages reach the SIP service provider interface (SPI). Setting **udp** or **tcp** as the protocol causes this to be the protocol for which SIP user agents 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** command.

## Examples

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

```
Router(config)# sip-ua
Router(config-sip-ua)# no transport tcp
```

## Related Commands

| Command | Description |
|---|---|
| **sip-ua** | Enables the SIP user-agent configuration commands, with which the user agent is configured. |

# trunk group

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. |

## Defaults

No default behavior or values

## Command Modes

Global configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(11)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.

## Examples

The following example assigns the number 5 to a trunk group profile:

```
Router(config)# trunk group 5
Router(config-trunk-group)# carrier-id allcalls
Router(config-trunk-group)# maxcalls 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 assigns the name “newyork” to a trunk group profile:

```
Router(config)# trunk group newyork
Router(config-trunk-group)# carrier-id local
Router(config-trunk-group)# maxcalls 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 (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). |

## Defaults

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). |

## Defaults

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

## Command Modes

Dial-peer configuration

## 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). |

## Defaults

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). |

## Defaults

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. |

## Defaults

No default behavior or values

## Command Modes

Dial-peer configuration

## 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. |

## Defaults

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. |

# 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). |

## Defaults

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 (ephone)

To define a phone type or to define one or two add-on phone modules for a Cisco IP phone, use the **type** command in ephone configuration mode. To remove a definition, use the **no** form of this command.

**type** *phone-type* [**addon 1** *module-type* [**2** *module-type*]]

**no type** *phone-type* [**addon 1** *module-type* [**2** *module-type*]]

## Syntax Description

| | |
|---|---|
| *phone-type* | Type of IP phone that is being defined or the type of IP phone to which a module is being added. Valid entries are:<br>• **7910**—Cisco IP Phone 7910<br>• **7935**—Cisco IP Conference Station 7935<br>• **7940**—Cisco IP Phone 7940<br>• **7960**—Cisco IP Phone 7960<br>• **ata**—Cisco ATA-186 or Cisco ATA-188<br>**Note** The only phones that accept an add-on module are the Cisco IP Phone 7940 and the Cisco IP Phone 7960. |
| **addon 1** *module-type* | (Optional) Tells the router that a module is being added to this IP phone, and the type of module. The valid entry for *module-type* is:<br>• **7914**—Cisco IP Phone Expansion Module 7914 |
| **2** *module-type* | (Optional) Tells the router that a second module is being added to this IP phone, and the type of module. The valid entry for *module-type* is:<br>• **7914**—Cisco IP Phone Expansion Module 7914 |

## Defaults

No default behavior or values

## Command Modes

Ephone configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(11)YT | This command was introduced. |
| 12.2(15)T | This command was integrated into Cisco IOS Release 12.2(15)T. |

## Usage Guidelines

Use this command with Cisco IOS Telephony Service (ITS) V2.1 or a later version.

In ITS V2.1, the only phones you are required to identify with the **type** command are the Cisco ATA-186 or the Cisco ATA-188. Cisco IOS software is able to identify all other phone types, but will take the configured value if found.

In ITS V2.1, the only phones that can accept an add-on module are the Cisco IP Phone 7940 and the Cisco IP Phone 7960.

## Examples

The following example defines an IP phone with tag 10 as a Cisco IP Phone 7960 with two attached Cisco IP Phone Expansion Module 7914s:

```
Router(config)# ephone 10
Router(config-ephone)# type 7960 addon 1 7914 2 7914
```

The following example defines the IP phone with tag 4 as a Cisco ATA device:

```
Router(config)# ephone 4
Router(config-ephone)# mac 1234.87655.234
Router(config-ephone)# type ata
```

## Related Commands

| Command | Description |
|---|---|
| **ephone** | Enters ephone configuration mode to register Cisco IP phones. |

# 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. |

## Defaults

**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. |
| **retry-limit** | Sets the connection retry limit. |

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

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

## Defaults

Type 1

## Command Modes

Voice-port configuration

## Command History

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

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

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

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