Cisco IOS Voice Commands: R

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*.

radius-server attribute 6

To set an option for RADIUS Attribute 6 (Service-Type) values in a RADIUS profile, use the **radius-server attribute 6** command in global configuration mode. To disable all options, use the **no** form of this command.

radius-server attribute 6 {**on-for-login-auth** | **support-multiple** | **voice 1**}

no radius-server attribute 6

Syntax Description

on-for-login-auth	Sends Attribute 6 (Service-Type) in the authentication packet.
support-multiple	Supports multiple service-type values in each RADIUS profile.
voice 1	Selects the service-type value for voice calls. The **voice 1** keyword pair sets the service-type value to login or login-user.

Defaults

None of the Attribute 6 options is enabled.

Command Modes

Global configuration

Command History

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

Usage Guidelines

The **support-multiple** keyword allows for multiple instances of the Service-Type attribute to be present in an Access-Accept packet. The default behavior is to disallow multiple instances, which results in treating an Access-Accept that contains them as though an Access-Reject was received.

Examples

The following example sets support for multiple service-type values in each RADIUS profile:

```
Router(config)# radius-server attribute 6 support-multiple
```

ras retry

To configure the H.323 Registration, Admission, and Status (RAS) message retry counters, use the **ras retry** command in voice service h323 configuration mode. To set the counters to the default values, use the **no** form of this command.

ras retry {**all** | **arq** | **brq** | **drq** | **grq** | **rai** | **rrq**} *value*

no ras retry {**all** | **arq** | **brq** | **drq** | **grq** | **rai** | **rrq**}

Syntax Description

all	Configures all RAS message counters that do not have explicit values configured individually. If **no ras retry all** is entered, all values are set to the default except for the individual values that were configured separately.
arq	Configures the admission request (ARQ) message counter.
brq	Configures the bandwidth request (BRQ) message counter.
drq	Configures the disengage request (DRQ) message counter.
grq	Configures the gatekeeper request (GRQ) message counter.
rai	Configures the resource availability indication (RAI) message counter.
rrq	Configures the registration request (RRQ) message counter.
value	Number of times for the gateway to resend messages to the gatekeeper after the timeout period. The timeout period is the period in which a message has not been received by the gateway from the gatekeeper and is configured using the **ras timeout** command. Valid values are 1 through 30.

Defaults

arq: 2 retries
brq: 2 retries
drq: 9 retries
grq: 2 retries
rai: 9 retries
rrq: 2 retries

Command Modes

Voice service h323 configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

Use this command in conjunction with the **ras timeout** command. The **ras timeout** command configures the number of seconds for the gateway to wait before resending a RAS message to a gatekeeper. The **ras retry** command configures the number of times to resend the RAS message after the timeout period expires. The default values for timeouts and retries are acceptable in most networks. You can use these commands if you are experiencing problems in RAS message transmission between gateways and gatekeepers. For example, if you have gatekeepers that are slow to respond to a type of RAS request, increasing the timeout value and the number of retries increases the call success rate, preventing lost billing information and unnecessary switchover to an alternate gatekeeper.

Examples

The following example shows the GRQ message counter set to 5 and all other RAS message counters set to 10:

```
Router(conf-serv-h323)# ras retry all 10
Router(conf-serv-h323)# ras retry grq 5
```

Related Commands

Command	Description
ras timeout	Configures the H.323 Registration, Admission, and Status (RAS) message timeout values.

ras rrq dynamic prefixes

To enable advertisement of dynamic prefixes in additive registration request (RRQ) RAS messages on the gateway, use the **ras rrq dynamic prefixes** command in voice service h323 configuration mode. To disable advertisement of dynamic prefixes in additive RRQ messages, use the **no** form of this command.

ras rrq dynamic prefixes

no ras rrq dynamic prefixes

Syntax Description

This command has no arguments or keywords.

Defaults

In Cisco IOS Release 12.2(15)T, the default was set to enabled. In Cisco IOS Release 12.3(3), the default is set to disabled.

Command Modes

Voice service h323 configuration

Command History

Release	Modification
12.2(15)T	This command was introduced.
12.3(3)	The default is modified to be disabled by default.

Usage Guidelines

In Cisco IOS Release 12.2(15)T, the default for the **ras rrq dynamic prefixes** command was set to enabled so that the gateway automatically sent dynamic prefixes in additive RRQ messages to the gatekeeper. Beginning in Cisco IOS Release 12.3(3), the default is set to disabled, and you must specify the command to enable the functionality.

Examples

The following example allows the gateway to send advertisements of dynamic prefixes in additive RRQ messages to the gatekeeper:

```
Router(conf-serv-h323)# ras rrq dynamic prefixes
```

Related Commands

Command	Description
rrq dynamic-prefixes-accept	Enables processing of additive RRQ messages and dynamic prefixes on the gatekeeper.

ras rrq ttl

To configure the H.323 Registration, Admission, and Status (RAS) registration request (RRQ) time-to-live value, use the **ras rrq ttl** command in voice service h323 configuration mode. To set the RAS RRQ time-to-live value to the default value, use the **no** form of this command.

ras rrq ttl *time-to-live* [**margin** *time*]

no ras rrq ttl

Syntax Description

time-to-live	Number of seconds that the gatekeeper should consider the gateway active. Valid values are 15 through 4000. The *time-to-live* value must be greater than the **margin** *time* value.
margin *time*	The number of seconds that an RRQ message can be transmitted from the gateway before the time-to-live value advertised to the gatekeeper. Valid values are 1 through 60. The **margin** *time* value times two must be less than or equal to the *time-to-live* value.

Defaults

time-to-live: 60 seconds
margin *time*: 15 seconds

Command Modes

Voice service h323 configuration

Command History

Release	Modification
12.3(1)	This command was introduced.
12.3(6)	The maximum *time-to-live* value was changed from 300 to 4000 seconds.

Usage Guidelines

Use this command to configure the number of seconds that the gateway should be considered active by the gatekeeper. The gateway transmits this value in the RRQ message to the gatekeeper. The **margin** *time* keyword and argument allow the gateway to transmit an early RRQ to the gatekeeper before the time-to-live value advertised to the gatekeeper.

Examples

The following example shows the time-to-live value configured to 300 seconds and the **margin** *time* value configured to 60 seconds:

```
Router(conf-serv-h323)# ras rrq ttl 300 margin 60
```

ras timeout

To configure the H.323 Registration, Admission, and Status (RAS) message timeout values, use the **ras timeout** command in voice service h323 configuration mode. To set the timers to the default values, use the **no** form of this command.

ras timeout {**all** | **arq** | **brq** | **drq** | **grq** | **rai** | **rrq**} *value*

no ras timeout {**all** | **arq** | **brq** | **drq** | **grq** | **rai** | **rrq**}

Syntax Description

all	Configures message timeout values for all RAS messages that do not have explicit values configured individually. If **no ras timeout all** is entered, all values are set to the default except for the individual values that were configured separately.
arq	Configures the admission request (ARQ) message timer.
brq	Configures the bandwidth request (BRQ) message timer.
drq	Configures the disengage request (DRQ) message timer.
grq	Configures the gatekeeper request (GRQ) message timer.
rai	Configures the resource availability indication (RAI) message timer.
rrq	Configures the registration request (RRQ) message timer.
value	Number of seconds for the gateway to wait for a message from the gatekeeper before timing out. Valid values are 1 through 45.

Defaults

arq: 3 seconds
brq: 3 seconds
drq: 3 seconds
grq: 5 seconds
rai: 3 seconds
rrq: 5 seconds

Command Modes

Voice service h323 configuration

Command History

Release	Modification
12.3(1)	This command was introduced.

Usage Guidelines

Use this command in conjunction with the **ras retry** command. The **ras timeout** command configures the number of seconds for the gateway to wait before resending a RAS message to a gatekeeper. The **ras retry** command configures the number of times to resend the RAS message after the timeout period expires. The default values for timeouts and retries are acceptable in most networks. You can use these commands if you are experiencing problems in RAS message transmission between gateways and gatekeepers. For example, if you have gatekeepers that are slow to respond to a type of RAS request, increasing the timeout value and the number of retries increases the call success rate, preventing lost billing information and unnecessary switchover to an alternate gatekeeper.

Examples

The following example shows the GRQ message timeout value set to 10 seconds and all other RAS message timeout values set to 7 seconds:

```
Router(conf-serv-h323)# ras timeout grq 10
Router(conf-serv-h323)# ras timeout all 7
```

Related Commands

Command	Description
ras retry	Configures the H.323 Registration, Admission, and Status (RAS) message retry counters.

rbs-zero

To enable 1AESS switch support for T1 lines on the primary serial interface of an access server, use the **rbs-zero** command in serial interface configuration mode. To disable IAESS switch support, use the **no** form of this command.

rbs-zero [**nfas-int** *nfas-int-range*]

no rbs-zero [**nfas-int** *nfas-int-range*]

Syntax Description

nfas-int *nfas-int-range*	(Optional) Non-Facility Associated Signaling (NFAS) interface number. Range is from 0 to 32.

Defaults

1AESS switch support is disabled.

Command Modes

Serial interface configuration

Command History

Release	Modification
12.2(2)XA	This command was introduced.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command was applicable to the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800 in this release.

Usage Guidelines

Use this command to configure the primary serial interface of an access server connected to T1 lines to support 1AESS switches for dial-in and dial-out calls. Modem calls of 56K or a lower rate are accepted; 64K calls are rejected.

In IAESS mode, the following occurs:

- Modem calls are accepted and digital calls are rejected.
- The ABCD bit of the 8 bits in the incoming calls is ignored. The ABCD bit of the 8 bits in the outgoing modem calls is set to 0.

In non-1AESS mode, modem and digital calls are accepted.

Examples

The following example enables 1AESS switching support on T1 channel 0:

```
Router(config)# controller t1 1/0
Router(config-controller)# framing esf
Router(config-controller)# linecode b8zs
Router(config-controller)# pri-group timeslots 1-24 nfas_d primary nfas_int 0 nfas_group 1
```

```
Router(config)# interface serial 1/0:23
Router(config-if)# no ip address
Router(config-if)# isdn switch-type primary-ni
Router(config-if)# rbs-zero nfas-int 0
```

Related Commands

Command	Description
interface serial	Enters serial interface configuration mode.
isdn switch-type	Sets the switch type.
pri-group timeslots	Configures the PRI trunk for a designated operation.
show controllers t1	Displays information about the T1 links and the hardware and software driver information for the T1 controller.
show isdn nfas group	Displays all the members of a specified NFAS group or all NFAS groups.

redirection (SIP)

To enable the handling of 3*xx* redirect messages, use the **redirection** command in SIP user agent configuration mode. To disable the handling of 3*xx* redirect messages, use the **no** form of this command.

redirection

no redirection

Syntax Description

This command has no arguments or keywords.

Defaults

Redirection is enabled.

Command Modes

SIP user agent configuration

Command History

Release	Modification
12.2(13)T	This command was introduced.

Usage Guidelines

The **redirection** command applies to all Session Initiation Protocol (SIP) VoIP dial peers configured on the gateway.

The default mode of SIP gateways is to process incoming 3*xx* redirect messages according to RFC 2543. However if redirect handling is disabled with the **no redirection** command, the gateway treats the incoming 3*xx* responses as 4*xx* error class responses. To reset the default processing of 3*xx* messages, use the **redirection** command.

Examples

The following example disables processing of incoming 3*xx* redirection messages:

```
Router(config)# sip-ua
Router(config-sip-ua)# no redirection
```

Related Commands

Command	Description
show sip-ua statistics	Displays response, traffic, and retry SIP statistics.
show sip-ua status	Displays SIP UA status.

register e164

To configure a gateway to register or deregister a fully-qualified dial-peer E.164 address with a gatekeeper, use the **register e164** command in dial-peer configuration mode. To deregister the E.164 address, use the **no** form of this command.

register e164

no register e164

Syntax Description

This command has no arguments or keywords.

Defaults

No E.164 addresses are registered until you enter this command.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(5)T	This command was introduced.
12.1(5)XM2	The command was implemented on the Cisco AS5350 and Cisco AS5400.
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. This command is supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400, and the Cisco AS5850 in this release.

Usage Guidelines

Use this command to register the E.164 address of an analog telephone line attached to a foreign exchange station (FXS) port on a router. The gateway automatically registers fully qualified E.164 addresses. Use the **no register e164** command to deregister an address. Use the **register e164** command to register a deregistered address.

Before you automatically or manually register an E.164 address with a gatekeeper, you must create a dial peer (using the **dial-peer** command), assign an FXS port to the peer (using the **port** command), and assign an E.164 address using the **destination-pattern** command. The E.164 address must be a fully qualified address. For example, +5551212, 5551212, and 4085551212 are fully qualified addresses; 408555.... is not. E.164 addresses are registered only for active interfaces, which are those that are not shut down. If an FXS port or its interface is shut down, the corresponding E.164 address is deregistered.

Tip

You can use the **show gateway** command to find out whether the gateway is connected to a gatekeeper and whether a fully qualified E.164 address is assigned to the gateway. Use the **zone-prefix** command to define prefix patterns on the gatekeeper, such as 408555...., that apply to one or more gateways.

Examples

The following command sequence places the gateway in dial-peer configuration mode, assigns an E.164 address to the interface, and registers that address with the gatekeeper.

```
gateway1(config)# dial-peer voice 111 pots
gateway1(config-dial-peer)# port 1/0/0
gateway1(config-dial-peer)# destination-pattern 5551212
gateway1(config-dial-peer)# register e164
```

The following commands deregister an address with the gatekeeper.

```
gateway1(config)# dial-peer voice 111 pots
gateway1(config-dial-peer)# no register e164
```

The following example shows that you must have a connection to a gatekeeper and must define a unique E.164 address before you can register an address.

```
gateway1(config)# dial-peer voice 222 pots
gateway1(config-dial-peer)# port 1/0/0
gateway1(config-dial-peer)# destination 919555....
gateway1(config-dial-peer)# register e164
ERROR-register-e164:Dial-peer destination-pattern is not a full E.164
number
gateway1(config-dial-peer)# no gateway
gateway1(config-dial-peer)# dial-peer voice 111 pots
gateway1(config-dial-peer)# register e164
ERROR-register-e164:No gatekeeper
```

Related Commands

Command	Description
destination-pattern	Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer.
dial-peer (voice)	Enters dial-peer configuration mode and specifies the method of voice encapsulation.
port (dial peer)	Associates a dial peer with a specific voice port.
show gateway	Displays the current gateway status.
zone prefix	Adds a prefix to the gatekeeper zone list.

registered-caller ring

To configure the Nariwake service registered caller ring cadence, use the **registered-caller ring** command in dial-peer configuration mode.

registered-caller ring *cadence*

Syntax Description

cadence	A value of 0, 1, or 2. The default ring cadence for registered callers is 1 and for unregistered callers is 0. The on and off periods of ring 0 (normal ringing signals) and ring 1 (ringing signals for the Nariwake service) are defined in the NTT user manual.

Defaults

The default Nariwake service registered caller ring cadence is ring 1.

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

If your ISDN line is provisioned for the I Number or dial-in services, you must also configure a dial peer by using the **destination-pattern not-provided** command. Either port 1 or port 2 can be configured under this dial peer. The router then forwards the incoming call to voice port 1. (See the “Examples” section below.

If more than one dial peer is configured with the **destination-pattern not-provided** command, the router uses the first configured dial peer for the incoming calls. To display the Nariwake ring cadence setting, use the **show run** command.

Examples

The following example sets the ring cadence for registered callers to 2.

```
pots country jp
dial-peer voice 1 pots
 registered-caller ring 2
```

rel1xx

To enable all Session Initiation Protocol (SIP) provisional responses (other than 100 Trying) to be sent reliably to the remote SIP endpoint, use the **rel1xx** command in SIP configuration mode. To reset to the default, use the **no** form of this command.

rel1xx {**supported** *value* | **require** *value* | **disable**}

no rel1xx

Syntax Description

supported *value*	Supports reliable provisional responses. The *value* argument may have any value, as long as both the user-agent client (UAC) and user-agent server (UAS) configure it the same. This keyword, with *value* of 100rel, is the default.
require *value*	Requires reliable provisional responses. The *value* argument may have any value, as long as both the UAC and UAS configure it the same.
disable	Disables the use of reliable provisional responses.

Defaults

supported with the 100rel value

Command Modes

SIP configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 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

The use of resource reservation with SIP requires that the reliable provisional feature for SIP be enabled either at the VoIP dial-peer level or globally on the router.

There are two ways to configure reliable provisional responses:

- Dial-peer mode. You can configure reliable provisional responses for the specific dial peer only by using the **voice-class sip rel1xx** command.
- SIP mode. You can configure reliable provisional responses globally by using the **rel1xx** command.

The **voice-class sip rel1xx** command in dial-peer configuration mode takes precedence over the **rel1xx** command in global configuration mode with one exception: If the **voice-class sip rel1xx** command is used with the **system** keyword, the gateway uses what was configured under the **rel1xx** command in global configuration mode.

Enter SIP configuration mode from voice-service VoIP configuration mode as shown in the following example.

Examples

The following example shows use of the **rel1xx** command with the value 100rel:

```
Router(config)# voice service voip
Router(config-voi-srv)# sip
Router(conf-serv-sip)# rel1xx supported 100rel
```

Related Commands

Command	Description
sip	Enters SIP configuration mode from voice-service VoIP configuration mode.
voice-class sip rel1xx	Provides provisional responses for calls on a dial peer basis.

remote-party-id

To enable translation of the SIP header Remote-Party-ID, use the **remote-party-id** command in SIP user agent configuration mode. To disable Remote-Party-ID translation, use the **no** form of this command.

remote-party-id

no remote-party-id

Syntax Description

remote-party-id	Extracts the calling name and number from the Remote-Party-ID header.
no remote-party-id	Extracts the calling name and number from the From header.

Defaults

Remote-Party-ID translation is enabled

Command Modes

SIP user agent configuration

Command History

Release	Modification
12.2(13)T	This command was introduced.

Usage Guidelines

When the **remote-party-id** command is enabled, one of the following calling information treatments occurs:

- If a Remote-Party-ID header is present in the incoming INVITE message, the calling name and number extracted from the Remote-Party-ID header are sent as the calling name and number in the outgoing Setup message. This is the default behavior. Use the remote-party-id command to enable this option.
- When no Remote-Party-ID header is available, no translation occurs so the calling name and number are extracted from the From header and are sent as the calling name and number in the outgoing Setup message. This treatment also occurs when the feature is disabled.

Examples

The following example shows the Remote-Party-ID translation being enabled:

```
Router(config-sip-ua)# remote-party-id
```

Related Commands

Command	Description
debug ccsip events	Enables tracing of SIP SPI events.
debug ccsip messages	Enables SIP SPI message tracing.
debug isdn q931	Displays call setup and teardown of ISDN connections.
debug voice ccapi in out	Enables tracing the execution path through the call control API.

req-qos

To specify the desired quality of service to be used in reaching a specified dial peer, use the **req-qos** command in dial-peer configuration mode. To reset to the default, use the **no** form of this command.

req-qos {**best-effort** | **controlled-load** | **guaranteed-delay**}

no req-qos

Syntax Description

best-effort	Resource Reservation Protocol (RSVP) makes no bandwidth reservation.
controlled-load	RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to assure that preferential service is received even when the bandwidth is overloaded.
guaranteed-delay	RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queueing if the bandwidth reserved is not exceeded.

Defaults

best-effort

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

This command is applicable only to VoIP dial peers.

Use this command to request a specific quality of service to be used in reaching a dial peer. Like **acc-qos**, when you issue this command, the Cisco IOS software reserves a certain amount of bandwidth so that the selected quality of service can be provided. Cisco IOS software uses Resource Reservation Protocol (RSVP) to request quality of service guarantees from the network.

Examples

The following example requests guaranteed-delay quality of service for a dial peer:

```
dial-peer voice 10 voip
 req-qos guaranteed-delay
```

Related Commands

Command	Description
acc-qos	Defines the acceptable QoS for any inbound and outbound call on a VoIP dial peer.

reset

To reset a set of digital signal processors (DSPs), use the **reset** command in global configuration mode.

reset *number*

Syntax Description

number	Number of DSPs to be reset. Range is from 0 to 30.

Defaults

No default behavior or values.

Command Modes

Global configuration

Command History

12.0(5)XE	This command was introduced on the Cisco 7200 series.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.

Examples

The following example displays the reset command configuration for DSP 1:

```
reset 1
01:24:54:%DSPRM-5-UPDOWN: DSP 1 in slot 1, changed state to up
```

reset (cm-fallback)

To reset Cisco IP phones, use the **reset** command in call-manager-fallback configuration mode.

reset {**all** *seconds* | **mac-address** *mac-address*}

Syntax Description

all	All the Cisco IP phones.
seconds	Time interval, in seconds, between each phone reset. Range is from 0 to 15. Default is 0.
mac-address *mac-address*	MAC address of a particular Cisco IP phone.

Defaults

0 seconds

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

Usage Guidelines

This command does not have a **no** form.

Examples

The following example resets all Cisco IP phones:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# reset all 5
```

The following example resets the Cisco IP phone with MAC address CFBA.321B.96FA:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# reset mac-address CFBA.321B.96FA
```

Related Commands

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

reset (ephone)

To reset the Cisco IP phones in ephone configuration mode, use the r**eset** command in ephone configuration mode.

reset

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Ephone configuration

Command History

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

Usage Guidelines

This command does not have a **no** form.

Examples

The following example resets the Cisco IP phones:

```
Router(config)# ephone 1
Router(config-ephone)# reset
```

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.

reset (telephony-service)

To perform a complete reboot of one or all phones associated with a Cisco IOS Telephony Service (ITS) router, use the **reset** command in telephony-service configuration mode. To interrupt and cancel a sequential reset cycle, use the **no** form of the command.

reset {**all** [*time-interval*] | **cancel** | **mac-address** *mac-address* | **sequence-all**}

no reset all

Syntax Description

all	All Cisco IP phones served by this ITS router. This keyword causes the router to pause 15 seconds between the reset start for each successive phone.
time-interval	Time interval, in seconds, between each phone reset. Range is 0 to 60. Default is 15.
cancel	Interrupts a sequential reset cycle.
mac-address *mac-address*	MAC address of a particular Cisco IP phone.
sequence-all	Resets all phones in strict one-at-a-time order by waiting for one phone to finish before starting the reset for the next phone. There is a reset timeout of 4 minutes, after which the router stops waiting for the currently registering phone to complete registration and starts to reset the next phone.

Defaults

time-interval is 15.

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 series.
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	This command was implemented on the Cisco 1760.
12.2(11)YT	The *time-interval* range maximum was increased from 15 to 60 and the default was changed from 0 to 15.
12.2(11)YT1	The **cancel** keyword and the **sequence-all** keyword were introduced.
12.2(15)T	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

After you update information for one or more phones associated with an ITS router, the phone or phones must be rebooted. There are two commands to reboot the phones: **reset** and **restart**. The **reset** command performs a “hard” reboot similar to a power-off-power-on sequence. It reboots the phone and sources the

DHCP server and TFTP server to update from their information as well. The **restart** command performs a “soft” reboot by simply rebooting the phone without contacting the DHCP and TFTP servers. The **reset** command takes significantly longer to process than the **restart** command when you are updating multiple phones, but it must be used to update phone firmware, user locale, network locale, or URL parameters. For simple button, line, or speed-dial changes, you can use the **restart** command.

When using the **reset** command, the default time interval of 15 seconds is recommended for an 8- to 10-phone office so that all the phones do not attempt to access TFTP server resources simultaneously. This value should be modified accordingly for larger networks.

When you use the **reset sequence-all** command, the router waits for one phone to complete its reset before starting to reset the next phone. The delay provided by this command prevents multiple phones attempting to access the TFTP server simultaneously and therefore failing to reset properly. Each reset operation can take several minutes when you use this command. There is a reset timeout of 4 minutes, after which the router stops waiting for the currently registering phone to complete registration and starts to reset the next phone.

The **reset sequence-all** command is required when the phone firmware version, user locale, or network locale is changed, and is automatically selected over the **reset all** command when any of those three parameters are changed. However, this automatic selection of the **reset sequence-all** command can be overridden by using the **reset all** *time-interval* command when the time interval is set to some value other than the default of 15 seconds.

To interrupt and terminate an ongoing sequential reset cycle, use the **reset cancel** command.

The **restart all** command allows the system to perform quick phone resets in which only the button template, line information, and speed-dial information is updated. Refer to the command reference entry for **restart all** for more information.

Examples

The following example resets all IP phones served by this ITS router:

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

The following example resets the Cisco IP phone with the MAC address CFBA.321B.96FA:

```
Router(config)# telephony-service
Router(config-telephony-service)# reset mac-address CFBA.321B.96FA
```

The following example resets all IP phones in sequential, non-overlapping order:

```
Router(config)# telephony-service
Router(config-telephony-service)# reset sequence-all
```

Related Commands

Command	Description
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco IOS Telephony Service (ITS) router.
restart all (telephony-service)	Performs a fast reboot of all phones associated with a Cisco IOS Telephony Service (ITS) router.
telephony-service	Enables Cisco ITS and enters telephony-service configuration mode.

resource threshold

To configure a gateway to report H.323 resource availability to its gatekeeper, use the **resource threshold** command in gateway configuration mode. To disable gateway resource-level reporting, use the **no** form of this command.

resource threshold [**all**] [**high** *percentage-value*] [**low** *percentage-value*] [**report-policy** {**idle-only** | **addressable**}]

no resource threshold

Syntax Description

all	(Optional) High- and low-parameter settings are applied to all monitored H.323 resources. This is the default condition.
high *percentage-value*	(Optional) Resource utilization level that triggers a Resource Availability Indicator (RAI) message that indicates that H.323 resource use is high. Enter a number between 1 and 100 that represents the high-resource utilization percentage. A value of 100 specifies high-resource usage when any H.323 resource is unavailable. Default is 90 percent.
low *percentage-value*	(Optional) Resource utilization level that triggers an RAI message that indicates H.323 resource usage has dropped below the high-usage level. Enter a number between 1 and 100 that represents the acceptable resource utilization percentage. After the gateway sends a high-utilization message, it waits to send the resource recovery message until the resource use drops below the value defined by the **low** parameter. Default is 90 percent.
report-policy	(Optional) Selects how resource utilization is calculated. Available resources to be reported can be one of the following: • **idle-only**—Includes free and in-use channels only. This is the default calculation. • **addressable**—Includes free, in-use and disabled channels.

Defaults

Reports low resources when 90 percent of resources are in use and reports resource availability when resource use drops below 90 percent.

Command Modes

Gateway configuration

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco AS5300.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
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.

Release	Modification
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. This command is supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release

Usage Guidelines

This command defines the resource load levels that trigger RAI messages. To view the monitored resources, enter the **show gateway** command.

The monitored H.323 resources include digital signal processor (DSP) channels and DS0s. Use the **show call resource voice stats** command to see the total amount of resources available for H.323 calls.

Note The DS0 resources that are monitored for H.323 calls are limited to the ones that are associated with a voice POTS dial peer.

See the **dial-peer** configuration commands for details on how to associate a dial peer with a PRI or channel-associated signaling (CAS) group.

When any monitored H.323 resources exceed the threshold level defined by the **high** parameter, the gateway sends an RAI message to the gatekeeper with the AlmostOutOfResources field flagged. This message reports high resource usage.

When all gateway H.323 resources drop below the level defined by the **low** parameter, the gateway sends the RAI message to the gatekeeper with the AlmostOutOfResources field cleared.

When a gatekeeper can choose between multiple gateways for call completion, the gatekeeper uses internal priority settings and gateway resource statistics to determine which gateway to use. When all other factors are equal, a gateway that has available resources is chosen over a gateway that has reported limited resources.

You can select which channels to include in threshold calculation.

- idle-only: Utilization = (In-use)/(In-use plus Free)
- addressable: Utilization = (In-use plus Disabled)/Addressable

The addressable channel calculation includes disabled channels and therefore provides a more accurate percentage of available channels.

Examples

The following example defines the H.323 resource limits for a gateway.

```
gateway1(config-gateway)# resource threshold high 70 low 60
```

Related Commands

Command	Description
show call resource voice stats	Displays resource statistics for an H.323 gateway.
show call resource voice threshold	Displays the threshold configuration settings and status for an H.323 gateway.
show gateway	Displays the current gateway status.

response-timeout

To configure the maximum time to wait for a response from a server, use the **response-timeout** command in settlement configuration mode. To reset to the default, use the **no** form of this command.

response-timeout *number*

no response-timeout *number*

Syntax Description

number	Response waiting time, in seconds. Default is 1.

Defaults

1 second

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

If no response is received within the response-timeout time limit, the current connection ends, and the router attempts to contact the next service point.

Examples

The following example sets response timeout to 1 second.

```
settlement 0
 response-timeout 1
```

Related Commands

Command	Description
connection-timeout	Configures the time for which a connection is maintained after completion of a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
retry-delay	Sets the time between attempts to connect with the settlement provider.
retry-limit	Sets the maximum number of attempts to connect to the provider.

Command	Description
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement mode and specifies the attributes specific to a settlement provider.
show settlement	Displays the configuration for all settlement server transactions.
shutdown/no shutdown	Deactivates the settlement provider/activates the settlement provider.
type	Configures an SAA-RTR operation type.
url	Specifies the Internet service provider address.

restart (ephone)

To perform a fast reboot of a single phone associated with a Cisco IOS Telephony Service (ITS) router after updating buttons, lines, or speed-dial numbers, use the **restart** command in ephone configuration mode. To cancel the reboot, use the **no** form of this command.

restart

no restart

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Ephone configuration

Command History

Release	Modification
12.2(11)YT1	This command was introduced.
12.2(15)T	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command causes the system to perform a fast phone reset in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the **reset** command. The **restart** command is much faster than the **reset** command because the phone does not need to access the DHCP or TFTP server.

To restart all phones in an ITS system for quick changes to buttons, lines, and speed-dial numbers, use the **restart all** command in telephony-service configuration mode.

Examples

The following example restarts the phone with tag 1:

```
Router(config)# ephone 1
Router(config-ephone)# restart
```

Related Commands

Command	Description
reset	Performs a complete reboot of one or all phones associated with a Cisco IOS Telephony Service (ITS) router.
restart all (telephony-service)	Performs a fast reboot of all phones associated with a Cisco IOS Telephony Service (ITS) router.

restart all (telephony-service)

To perform a fast reboot of all phones associated with a Cisco IOS Telephony Service (ITS) router after updating buttons, lines, or speed-dial numbers, use the **restart all** command in telephony-service configuration mode. To cancel the reboot, use the **no** form of this command.

restart all

no restart all

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(11)YT1	This command was introduced.
12.2(15)T	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command causes the system to perform a fast phone reset in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the **reset** command.

Use the **restart all** command to reboot IP phones after quick changes to buttons, lines, and speed-dial numbers. It is much faster than the **reset** command because the phone does not access the DHCP or TFTP server.

To restart a single phone, use the **restart** command in ephone configuration mode.

Examples

The following example performs a quick restart of all phones in the ITS system:

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

Related Commands

Command	Description
reset	Performs a complete reboot of one or all phones associated with a Cisco IOS Telephony Service (ITS) router.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco IOS Telephony Service (ITS) router.

retry bye

To configure the number of times that a BYE request is retransmitted to the other user agent, use the **retry bye** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry bye *number*

no retry bye *number*

Syntax Description

number	Number of BYE retries. Range is from 1 to 10. The default is 10.

Defaults

10 retries

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms 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

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

Examples

The following example sets the number of BYE retries to 5.

```
sip-ua
 retry bye 5
```

Related Commands

Command	Description
default	Resets the value of a command to its default.
retry cancel	Configures the number of times that a CANCEL request is retransmitted to the other user agent.

Command	Description
retry comet	Configures the number of times that a COMET request is retransmitted to the other user agent.
retry invite	Configures the number of times that a Session Initiation Protocol (SIP) INVITE request is retransmitted to the other user agent.
retry notify	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
retry prack	Configures the number of times that the PRACK request is retransmitted to the other user agent.
retry rel1xx	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
retry response	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
sip-ua	Enables the SIP user-agent configuration commands, with which you configure the user agent.

retry cancel

To configure the number of times that a CANCEL request is retransmitted to the other user agent, use the **retry cancel** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry cancel *number*

no retry cancel *number*

Syntax Description

number	Number of CANCEL retries. Range is from 1 to 10. Default is 10.

Defaults

10 retries

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms 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

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

Examples

The following example sets the number of cancel retries to 5.

```
sip-ua
 retry cancel 5
```

Related Commands

Command	Description
default	Resets the value of a command to its default.
retry bye	Configures the number of times that a BYE request is retransmitted to the other user agent.

Command	Description
retry comet	Configures the number of times that a COMET request is retransmitted to the other user agent.
retry invite	Configures the number of times that a Session Initiation Protocol (SIP) INVITE request is retransmitted to the other user agent.
retry notify	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
retry prack	Configures the number of times that the PRACK request is retransmitted to the other user agent.
retry rel1xx	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
retry response	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
sip-ua	Enables the sip-ua configuration commands, with which you configure the user agent.

retry comet

To configure the number of times that a COMET request is retransmitted to the other user agent, use the **retry comet** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry comet *number*

no retry comet

Syntax Description

number	Number of COMET retries. Range is from 1 to 10. Default is 10.

Defaults

10 retries

Command Modes

SIP user-agent configuration

Command History

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

Usage Guidelines

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

Use the default number of 10 retries, when possible. Lower values, such as 1, can lead to an increased chance of the message not being received by the other user agent.

Examples

The following example configures a COMET request to be retransmitted 8 times:

```
Router(config)# sip-ua
Router(config-sip-ua)# retry comet 8
```

Related Commands

Command	Description
retry bye	Configures the number of times that a BYE request is retransmitted to the other user agent.
retry cancel	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
retry invite	Configures the number of times that a Session Initiation Protocol (SIP) INVITE request is retransmitted to the other user agent.
retry notify	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
retry prack	Configures the number of times that the PRACK request is retransmitted to the other user agent.
retry rel1xx	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
retry response	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
show sip-ua retry	Displays the SIP retry attempts.
show sip-ua statistics	Displays response, traffic, timer, and retry statistics.

retry interval

To define the time between border element attempts delivery of unacknowledged call-detail-record (CDR) information, use the **retry interval** command in Annex g neighbor service configuration mode. To reset to the default, use the **no** form of this command.

retry interval *seconds*

no retry interval

Syntax Description

seconds	Retry interval between delivery attempts, in seconds. Range is from 1 to 3600(1 hour). The default is 900.

Defaults

900 seconds

Command Modes

Annex g neighbor service configuration

Command History

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

Usage Guidelines

Use this command to set the interval during which the border element attempts delivery of unacknowledged call-detail-record (CDR) information.

Examples

The following example sets the retry interval to 2700 seconds (45 minutes):

```
Router(config-nxg-neigh-usg)# retry interval 2700
```

Related Commands

Command	Description
access-policy	Requires that a neighbor be explicitly configured.
inbound ttl	Sets the inbound time-to-live value.
outbound retry-interval	Defines the retry period for attempting to establish the outbound relationship between border elements.
retry window	Defines the total time for which a border element attempts delivery.
service-relationship	Establishes a service relationship between two border elements.
shutdown	Enables or disables the border element.
usage-indication	Enters the mode used to configure optional usage indicators.

retry invite

To configure the number of times that a Session Initiation Protocol (SIP) INVITE request is retransmitted to the other user agent, use the **retry invite** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry invite *number*

no retry invite *number*

Syntax Description

number	Number of INVITE retries. Range is from 1 to 10. Default is 6.

Defaults

6 retries

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms 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

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

When configuring SIP using SIP user-agent configuration commands such as the **retry invite** command, the use of the default values for the commands causes the rotary function to not take effect. The rotary function is when you set up more than one VoIP dial peer for the same destination pattern, and the dial peers are assigned to different targets. Assign different targets so that if the call cannot be set up with the first dial peer (preference one), the next dial peer can be tried.

To use the rotary function within SIP, set the retry value for the SIP **retry invite** command to 4 or less.

Examples

The following example sets the number of invite retries to 5.

```
sip-ua
 retry invite 5
```

Related Commands

Command	Description
default	Resets the value of a command to its default.
retry bye	Configures the number of times that a BYE request is retransmitted to the other user agent.
retry cancel	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
retry comet	Configures the number of times that a COMET request is retransmitted to the other user agent.
retry notify	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
retry prack	Configures the number of times that the PRACK request is retransmitted to the other user agent.
retry rel1xx	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
retry response	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
sip-ua	Enables the sip-ua configuration commands, with which you configure the user agent.

retry notify

To configure the number of times that the notify message is retransmitted to the user agent that initiated the transfer or Refer request, use the **retry notify** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry notify *number*

no retry notify

Syntax Description

number	Number of notify message retries. Range is from 1 to 10. Default is 10.

Defaults

10 retries

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 the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms 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

A notify message informs the user agent that initiated the transfer or refer request of the outcome of the Session Initiation Protocol (SIP) transaction.

Use the default number of 10 when possible. Lower values such as 1 can lead to an increased chance of the message not being received by the other user agent.

Examples

The following example configures a notify message to be retransmitted 10 times:

```
Router(config)# sip-ua
Router(config-sip-ua)# retry notify 10
```

Related Commands

Command	Description
retry bye	Configures the number of times that a BYE request is retransmitted to the other user agent.
retry cancel	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
retry comet	Configures the number of times that a COMET request is retransmitted to the other user agent.
retry invite	Configures the number of times that a Session Initiation Protocol (SIP) INVITE request is retransmitted to the other user agent.
retry prack	Configures the number of times that the PRACK request is retransmitted to the other user agent.
retry rel1xx	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
retry response	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
show sip-ua retry	Displays the SIP retry attempts.
show sip-ua statistics	Displays response, traffic, timer, and retry statistics.
timers notify	Sets the amount of time that the user agent should wait before retransmitting the Notify message.

retry prack

To configure the number of times that the PRACK request is retransmitted to the other user agent, use the **retry prack** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry prack *number*

no retry prack

Syntax Description

number	Number of PRACK retries. Range is from 1 to 10. Default is 10.

Defaults

10 retries

Command Modes

SIP user-agent configuration

Command History

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

Usage Guidelines

PRACK allows reliable exchanges of Session Initiation Protocol (SIP) provisional responses between SIP endpoints. Use the default number of 10 when possible. Lower values such as 1 can lead to an increased chance of the message not being received by the other user agent.

Examples

The following example configures a PRACK request to be retransmitted 9 times:

```
Router(config)# sip-ua
Router(config-sip-ua)# retry prack 9
```

Related Commands

Command	Description
retry bye	Configures the number of times that a BYE request is retransmitted to the other user agent.
retry cancel	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
retry comet	Configures the number of times that a COMET request is retransmitted to the other user agent.

Command	Description
retry invite	Configures the number of times that a Session Initiation Protocol (SIP) INVITE request is retransmitted to the other user agent.
retry notify	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
retry rel1xx	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
retry response	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
retry response	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
show sip-ua retry	Displays the SIP retry attempts.
show sip-ua statistics	Displays response, traffic, timer, and retry statistics.

retry refer

To configure the number of times that the Refer request is retransmitted, use the **retry refer** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry refer *number*

no retry refer

Syntax Description

number	Number of Refer request retries. Range is from 1 to 10. Default is 10.

Defaults

10 retries

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 Session Initiation Protocol (SIP) Refer request is sent by the originating gateway to the receiving gateway and initiates call forward and call transfer capabilities.

When configuring the **retry refer** command, use the default number of 10 when possible. Lower values such as 1 can lead to an increased chance of the message not being received by the receiving gateway.

Examples

The following example configures a Refer request to be retransmitted 10 times:

```
Router(config)# sip-ua
Router(config-sip-ua)# retry refer 10
```

Related Commands

Command	Description
show sip-ua retry	Displays the SIP retry attempts.
show sip-ua statistics	Displays response, traffic, timer, and retry statistics.

retry rel1xx

To configure the number of times that the reliable 1xx response is retransmitted to the other user agent, use the **retry rel1xx** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry rel1xx *number*

no retry rel1xx

Syntax Description

number	Number of reliable 1*xx* retries. Range is from 1 to 10. Default is 6.

Defaults

6 retries

Command Modes

SIP user-agent configuration

Command History

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

Usage Guidelines

Use the default number of 6 when possible. Lower values such as 1 can lead to an increased chance of the message not being received by the other user agent.

Examples

The following example configures the reliable 1*xx* response to be retransmitted 7 times:

```
Router(config)# sip-ua
Router(config-sip-ua)# retry rel1xx 7
```

Related Commands

Command	Description
retry bye	Configures the number of times that a BYE request is retransmitted to the other user agent.
retry cancel	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
retry comet	Configures the number of times that a COMET request is retransmitted to the other user agent.

Command	Description
retry invite	Configures the number of times that a Session Initiation Protocol (SIP) INVITE request is retransmitted to the other user agent.
retry notify	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
retry prack	Configures the number of times the PRACK request is retransmitted.
retry response	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
show sip-ua retry	Displays the SIP retry attempts.
show sip-ua statistics	Displays response, traffic, timer, and retry statistics.

retry response

To configure the number of times that the response message is retransmitted to the other user agent, use the **retry response** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry response *number*

no retry response

Syntax Description

number	Number of response retries. Range is from 1 to 10. Default is 6.

Defaults

6 retries

Command Modes

SIP user-agent configuration

Command History

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

Usage Guidelines

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

Examples

The following example sets the number of response retries to 5.

```
sip-ua
 retry response 5
```

Related Commands

Command	Description
default	Resets the value of a command to its default.
retry bye	Configures the number of times that a BYE request is retransmitted to the other user agent.
retry cancel	Configures the number of times that a CANCEL request is retransmitted to the other user agent.

Command	Description
retry comet	Configures the number of times that a COMET request is retransmitted to the other user agent.
retry invite	Configures the number of times that a Session Initiation Protocol (SIP) INVITE request is retransmitted to the other user agent.
retry notify	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
retry prack	Configures the number of times the PRACK request is retransmitted.
retry rel1xx	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
sip-ua	Enables the sip-ua configuration commands, with which you configure the user agent.

retry window

To define the total time for which a border element attempts delivery, use the **retry window** command in Annex G neighbor usage configuration mode. To reset to the default, use the **no** form of this command.

retry window *window-value*

no retry window

Syntax Description

window-value	Window value, in minutes. Range is from 1 to 65535. Default is 1440 minutes (24 hours).

Defaults

1440 minutes (24 hours)

Command Modes

Annex G neighbor usage configuration

Command History

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

Usage Guidelines

Use this command to set the total time during which a border element attempts delivery of unacknowledged call-detail-record (CDR) information.

Examples

The following example sets the retry window to 15 minutes:

```
Router(config-nxg-neigh-usg)# retry window 15
```

Related Commands

Command	Description
access-policy	Requires that a neighbor be explicitly configured.
inbound ttl	Sets the inbound time-to-live value.
outbound retry-interval	Defines the retry period for attempting to establish the outbound relationship between border elements.
retry bye	Configures the number of times that a BYE request is retransmitted to the other user agent.
retry cancel	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
retry comet	Configures the number of times that a COMET request is retransmitted to the other user agent.
retry invite	Configures the number of times that a Session Initiation Protocol (SIP) INVITE request is retransmitted to the other user agent.

Command	Description
retry notify	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
retry prack	Configures the number of times that the PRACK request is retransmitted to the other user agent.
retry rel1xx	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
retry response	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
service-relationship	Establishes a service relationship between two border elements.
shutdown	Enables or disables the border element.
usage-indication	Enters the submode used to configure optional usage indicators.

retry-delay

To set the time between attempts to connect with the settlement provider, use the **retry-delay** command in settlement configuration mode. To reset to the default, use the **no** form of this command.

retry-delay *number*

no retry-delay

Syntax Description

number	Interval, in seconds, between attempts to connect with the settlement provider. Range is from 1 to 600.

Defaults

2 seconds

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

After exhausting all service points for the provider, the router is delayed for the specified length of time before resuming connection attempts.

Examples

The following example sets a retry value of 15 seconds:

```
settlement 0
 relay-delay 15
```

Related Commands

Command	Description
connection-timeout	Configures the time for which a connection is maintained after completion of a communication exchange.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.
retry-limit	Sets the maximum number of attempts to connect to the provider.

Command	Description
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement configuration mode and specifies the attributes specific to a settlement provider.
show settlement	Displays the configuration for all settlement server transactions.
shutdown/no shutdown	Deactivates the settlement provider/activates the settlement provider.
type	Configures an SAA-RTR operation type.

retry-limit

To set the maximum number of attempts to connect to the provider, use the **retry-limit** command in settlement configuration mode. To reset to the default, use the **no** form of this command.

retry-limit *number*

no retry-limit *number*

Syntax Description

number	Maximum number of connection attempts in addition to the first attempt. Default is 1.

Defaults

1 retry

Command Modes

Settlement configuration

Command History

Release	Modification
12.0(4)XH1	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.

Usage Guidelines

If no connection is established after the configured number of retries has been attempted, the router ceases connection attempts. The retry limit number does not count the initial connection attempt. A retry limit of one (default) results in a total of two connection attempts to every service point.

Examples

The following example sets the number of retries to 1:

```
settlement 0
 retry-limit 1
```

Related Commands

Command	Description
connection-timeout	Configures the time for which a connection is maintained after a communication exchange is complete.
customer-id	Identifies a carrier or ISP with a settlement provider.
device-id	Specifies a gateway associated with a settlement provider.
encryption	Sets the encryption method to be negotiated with the provider.
max-connection	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
response-timeout	Configures the maximum time to wait for a response from a server.

Command	Description
retry-delay	Sets the time between attempts to connect with the settlement provider.
session-timeout	Sets the interval for closing the connection when there is no input or output traffic.
settlement	Enters settlement mode and specifies the attributes specific to a settlement provider.
show settlement	Displays the configuration for all settlement server transactions.
shutdown	Brings up the settlement provider.
type	Configures an SAA-RTR operation type.

ring

To set up a distinctive ring for your connected telephones, fax machines, or modems, use the **ring** command in interface configuration mode. To disable the ring, use the **no** form of this command.

ring *cadence-number*

no ring *cadence-number*

Syntax Description

cadence-number	Number that determines the ringing cadence. Range is from 0 to 2: • Type 0 is a primary ringing cadence—default ringing cadence for the country your router is in. • Type 1 is a distinctive ring—0.8 seconds on, 0.4 seconds off, 0.8 seconds on, 0.4 seconds off. • Type 2 is a distinctive ring—0.4 seconds on, 0.2 seconds off, 0.4 seconds on, 0.2 seconds off, 0.8 seconds on, 4 seconds off.

Defaults

0

Command Modes

Interface configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco 800 series.

Usage Guidelines

This command applies to Cisco 800 series routers.

You can specify this command when creating a dial peer. This command does not work if it is not specified within the context of a dial peer. For information on creating a dial peer, refer to the *Cisco 800 Series Routers Software Configuration Guide*.

Examples

The following example specifies the type 1 distinctive ring:

```
ring 1
```

Related Commands

Command	Description
destination-pattern	Specifies the prefix, the full E.164 telephone number, or an ISDN directory number to be used for a dial peer.
dial-peer voice	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
no call-waiting	Disables call waiting.

Command	Description
port (dial-peer)	Enables an interface on a PA-4R-DTR port adapter to operate as a concentrator port.
pots distinctive-ring-guard-time	Specifies a delay during which a telephone port can be rung after a previous call is disconnected (for Cisco 800 series routers).
show dial-peer voice	Displays configuration information and call statistics for dial peers.

ring cadence

To specify the ring cadence for a Foreign Exchange Station (FXS) voice port, use the **ring cadence** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

ring cadence {*pattern-number* | **define** *pulse interval*}

no ring cadence

Syntax Description

pattern-number	Predefined ring cadence patterns. Each pattern specifies a ring-pulse time and a ring-interval time. • **pattern01**—2 seconds on, 4 seconds off • **pattern02**—1 second on, 4 seconds off • **pattern03**—1.5 seconds on, 3.5 seconds off • **pattern04**—1 second on, 2 seconds off • **pattern05**—1 second on, 5 seconds off • **pattern06**—1 second on, 3 seconds off • **pattern07**—0.8 second on, 3.2 seconds off • **pattern08**—1.5 seconds on, 3 seconds off • **pattern09**—1.2 seconds on, 3.7 seconds off • **pattern09**—1.2 seconds on, 4.7 seconds off • **pattern11**—0.4 second on, 0.2 second off, 0.4 second on, 2 seconds off • **pattern12**—0.4 second on, 0.2 second off, 0.4 second on, 2.6 seconds off
define	User-definable ring cadence pattern. Each number pair specifies one ring-pulse time and one ring-interval time. You must enter numbers in pairs, and you can enter from 1 to 6 pairs. The second number in the last pair that you enter specifies the interval between rings.
pulse	Number (1 or 2 digits) specifying ring-pulse (on) time in hundreds of milliseconds. Range is from 1 to 50, for pulses of 100 to 5000 ms. For example: 1 = 100 ms; 10 = 1 s, 40 = 4 s.
interval	Number (1 or 2 digits) specifying ring-interval (off) time in hundreds of milliseconds. Range is from 1 to 50, for pulses of 100 to 5000 ms. For example: 1 = 100 ms; 10 = 1 s, 40 = 4 s.

Defaults

Ring cadence defaults to the pattern that you specify with the **cptone** command.

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

The **patternXX** keyword provides preset ring cadence patterns for use on any platform. The **define** keyword allows you to create a custom ring cadence. On the Cisco 2600 and Cisco 3600 series routers, only one or two pairs of digits can be entered under the **define** keyword.

Examples

The following example sets the ring cadence to 1 second on and 4 seconds off on voice port 1/1 on a Cisco MC3810:

```
voice-port 1/1
 ring cadence pattern02
```

The following example sets the ring cadence to 1 second on, 1 second off, 1 second on, and 5 seconds off on voice port 1/2 on a Cisco MC3810:

```
voice-port 1/2
 ring cadence define 10 10 10 50
```

The following example sets the ring cadence to 1 second on and 2 seconds off on voice port 1/0/0 on a Cisco 2600 or Cisco 3600 series router:

```
voice-port 1/0/0
 ring cadence pattern04
```

Related Commands

Command	Description
cptone	Specifies the default tone, ring, and cadence settings according to country.
ring frequency	Specifies the ring frequency for a specified FXS voice port.

ring frequency

To specify the ring frequency for a specified Foreign Exchange Station (FXS) voice port, use the **ring frequency** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

ring frequency *number*

no ring frequency *number*

Syntax Description

number	Ring frequency, in hertz, used in the FXS interface. Valid entries are as follows: • Cisco 3600 series: 25 and 50. Default is 25. • Cisco MC3810: 20 and 30. Default is 20.

Defaults

Cisco 3600 series routers: 25 Hz
Cisco MC3810: 20 Hz

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

Use this command to select a specific ring frequency for an FXS voice port. Use the **no** form of this command to reset the default value. The ring frequency you select must match the connected equipment. If set incorrectly, the attached phone might not ring or might buzz. In addition, the ring frequency is usually country-dependent. You should take into account the appropriate ring frequency for your area before configuring this command.

This command does not affect ringback, which is the ringing a user hears when placing a remote call.

Examples

The following example sets the ring frequency on the Cisco 3600 series to 25 Hz:

```
voice-port 1/0/0
 ring frequency 25
```

The following example sets the ring frequency on the Cisco MC3810 to 20 Hz:

```
voice-port 1/1
 ring frequency 20
```

Related Commands

Command	Description
ring cadence	Specifies the ring cadence for an FXS voice port on the Cisco MC3810.
ring number	Specifies the number of rings for a specified FXO voice port.

ring number

To specify the number of rings for a specified Foreign Exchange Office (FXO) voice port, use the **ring number** command in voice-port configuration mode. To reset to the default, use the **no** form of this command.

ring number *number*

no ring number *number*

Syntax Description

number	Number of rings detected before answering the call. Range is from 1 to 10. The default is 1.

Defaults

1 ring

Command Modes

Voice-port configuration

Command History

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

Usage Guidelines

Use this command to set the maximum number of rings to be detected before answering a call over an FXO voice port. Use the **no** form of this command to reset the default value, which is one ring.

Normally, this command should be set to the default so that incoming calls are answered quickly. If you have other equipment available on the line to answer incoming calls, you might want to set the value higher to give the equipment sufficient time to respond. In that case, the FXO interface would answer if the equipment online did not answer the incoming call in the configured number of rings.

This command is not applicable to Foreign Exchange Station (FXS) or E&M interfaces because they do not receive ringing on incoming calls.

Examples

The following example on the Cisco 3600 series sets 5 as the maximum number of rings to be detected before closing a connection over this voice port:

```
voice-port 1/0/0
 ring number 5
```

The following example on the Cisco MC3810 sets 5 s as the maximum number of rings to be detected before closing a connection over this voice port:

```
voice-port 1/1
 ring number 5
```

Related Commands

Command	Description
ring frequency	Specifies the ring frequency for a specified FXS voice port.

roaming (dial-peer)

To enable roaming capability for a dial peer, use the **roaming** command in dial-peer configuration mode. To disable roaming capability, use the **no** form of this command.

roaming

no roaming

Syntax Description

This command has no arguments or keywords.

Defaults

No roaming

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

Use this command to enable roaming capability of a dial peer if that dial peer can terminate roaming calls. If a dial peer is dedicated to local calls only, disable roaming capability.

The roaming dial peer must work with a roaming service provider. If the dial peer allows a roaming user to go through and the service provider is not roaming-enabled, the call fails.

Examples

The following example enables roaming capability for a dial peer:

```
dial-peer voice 10 voip
roaming
```

Related Commands

Command	Description
roaming (settlement)	Enables the roaming capability for a settlement provider.
settle-call	Limits the dial peer to using only the specific clearinghouse identified by the specified *provider-number*.
settlement roam-pattern	Configures a pattern to match against when determining roaming.

roaming (settlement)

To enable roaming capability for a settlement provider, use the **roaming** command in settlement configuration mode. To disable roaming capability, use the **no** form of this command.

roaming

no roaming

Syntax Description

This command has no arguments or keywords.

Defaults

No roaming

Command Modes

Settlement configuration

Command History

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

Usage Guidelines

Enable roaming capability of a settlement provider if that provider can authenticate a roaming user and route roaming calls.

A roaming call is successful only if both the settlement provider and the outbound dial peer for that call are roaming-enabled.

Examples

The following example enables roaming capability for a settlement provider:

```
settlement 0
roaming
```

Related Commands

Command	Description
roaming (dial-peer mode)	Enables the roaming capability for the dial peer.
settle-call	Limits the dial peer to using only the specific clearinghouse identified by the specified *provider-number*.
settlement roam-pattern	Configures a pattern to match against when determining roaming.

rrq dynamic-prefixes-accept

To enable processing of additive registration request (RRQ) RAS messages and dynamic prefixes on the gatekeeper, use the **rrq dynamic-prefixes-accept** command in gatekeeper configuration mode. To disable processing of additive RRQ messages and dynamic prefixes, use the **no** form of this command.

rrq dynamic-prefixes-accept

no rrq dynamic-prefixes-accept

Syntax Description

This command has no arguments or keywords.

Defaults

In Cisco IOS Release 12.2(15)T, the default was set to enabled. In Cisco IOS Release 12.3(3), the default is set to disabled.

Command Modes

Gatekeeper configuration

Command History

Release	Modification
12.2(15)T	This command was introduced.
12.3(3)	The default is modified to be disabled by default.

Usage Guidelines

In Cisco IOS Release 12.2(15)T, the default for the **rrq dynamic-prefixes-accept** command was set to enabled so that the gatekeeper automatically received dynamic prefixes in additive RRQ messages from the gateway. Beginning in Cisco IOS Release 12.3(3), the default is set to disabled, and you must specify the command to enable the functionality.

Examples

The following example allows the gatekeeper to process additive RRQ messages and dynamic prefixes from the gateway:

```
Router(config-gk)# rrq dynamic-prefixes-accept
```

Related Commands

Command	Description
ras rrq dynamic prefixes	Enables advertisement of dynamic prefixes in additive RRQ messages on the gateway.

rtp payload-type

To identify the payload type of a Real-Time Transport Protocol (RTP) packet, use the **rtp payload-type** command in dial-peer configuration mode. To remove the RTP payload type, use the **no** form of this command.

rtp payload-type {cisco-cas-payload *number*| **cisco-clear-channel** *number* | **cisco-codec-fax-ack** *number* | **cisco-codec-fax-ind** *number* | **cisco-fax-relay** *number* | **cisco-pcm-switch-over-alaw** *number* | **cisco-pcm-switch-over-ulaw** *number*| **cisco-rtp-dtmf-relay** *number* | **nte** *number* | **nse** *number***} [comfort-noise {13 | 19}]**

no rtp payload-type nte

Syntax Description

cisco-cas-payload *number*	Cisco CAS RTP payload. Range is from 96 to 127. Default is 101.
cisco-clear-channel *number*	Cisco clear-channel RTP payload. Range is from 96 to 127. Default is 101.
cisco-codec-fax-ack *number*	Cisco codec fax acknowledge. Range is from 96 to 127. Default is 101.
cisco-codec-fax-ind *number*	Cisco codec fax indication. Range is from 96 to 127. Default is 101.
cisco-fax-relay *number*	Cisco fax relay. Range is from 96 to 127. Default is 101.
cisco-pcm-switch-over-alaw *number*	Cisco RTP PCM codec switch over indication (a-law). Range is from 96 to 127. Default is 101.
cisco-pcm-switch-over-ulaw *number*	Cisco RTP PCM codec switch over indication (u-law). Range is from 96 to 127. Default is 101.
cisco-rtp-dtmf-relay *number*	Cisco RTP DTMF relay. Range is from 96 to 127. Default is 101.
nte *number*	A named telephone event (NTE). Range is from 96 to 127. Default is 101.
nse *number*	A named signaling event (NSE). Range is from 96 to 127. Default is 101.
comfort-noise	(Optional) RTP payload type of comfort noise. The July 2001 draft entitled *RTP Payload for Comfort Noise*, from the IETF AVT working group, designates 13 as the payload type for comfort noise. Previous Cisco equipment uses 19 as the payload type for comfort noise. If you are connecting to a gateway that complies with the *RTP Payload for Comfort Noise* draft, use 13. Use 19 only if you are connecting to older Cisco gateways that use DSPware before version 3.4.32.

Defaults

101

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(2)T	This command was introduced.
12.2(2)XB	The **nte** and **comfort-noise** keywords were introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

Use this command to identify the payload type of an RTP NTE. Use this command after the **dtmf-relay** command is used to choose the NTE method of dual-tone multifrequency (DTMF) relay for a Session Initiation Protocol (SIP) call.

Examples

The following example identifies the RTTP payload type as NTE 99:

```
Router(config-dial-peer)# rtp payload-type nte 99
```

Related Commands

Command	Description
dtmf-relay	Specifies how an H.323 or SIP gateway relays DTMF tones between telephony interfaces and an IP network.

rtsp client session history duration

To specify how long to keep Real Time Streaming Protocol (RTSP) client history records in memory, use the **rtsp client session history duration** command in global configuration mode. To reset to the default, use the **no** form of this command.

rtsp client session history duration *minutes*

no rtsp client session history duration

Syntax Description

minutes	Duration, in minutes, to keep the record. Range is from 1 to 10000. Default is 10.

Defaults

10 minutes

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco AS5300.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	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(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751. This release does not support any other Cisco platforms.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400 and Cisco AS5850 in this release.

Examples

The following example sets the duration for the RTSP session history to 500 minutes:

```
rtsp client session history duration 500
```

Related Commands

Command	Description
call application voice load	Allows reload of an application that was loaded via the MGCP scripting package.
rtsp client session history records	Specifies the number of RTSP client session history records kept during the session.
show call application voice	Displays all TCL or MGCP scripts that are loaded.
show rtsp client session	Displays cumulative information about the RTSP session records.

rtsp client session history records

To configure the number of records to keep in the Real Time Streaming Protocol (RTSP) client session history, use the **rtsp client session history records** command in global configuration mode. To reset to the default, use the **no** form of this command.

rtsp client session history records *number*

no rtsp client session history records *number*

Syntax Description

number	Number of records to retain in a session history. Range is from 1 to 100000. Default is 50.

Defaults

50 records

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco AS5300.
12.1(5)T	This command was implemented on the Cisco AS5800.
12.1(5)XM2	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(4)XM	This command was implemented on the Cisco 1750 and Cisco 1751. This release does not support any other Cisco platforms.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 7200 series. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 is not included in this release.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400 and Cisco AS5850 in this release.

Examples

The following example specifies that a total of 500 records are to be kept in the RTSP client history:

```
rtsp client session history records 500
```

Related Commands

Command	Description
call application voice load	Allows reload of an application that was loaded via the MGCP scripting package.
rtsp client session history duration	Specifies the how long the RTSP is kept during the session.
show call application voice	Displays all TCL or MGCP scripts that are loaded.

rtsp client timeout connect

To set the number of seconds allowed for the router to establish a TCP connection to a Real -Time Streaming Protocol (RTSP) server, use the **rtsp client timeout connect** command in global configuration mode. To reset to the default, use the **no** form of this command.

rtsp client timeout connect *seconds*

no rtsp client timeout connect

Syntax Description

seconds	How long, in seconds, the router waits to connect to the server before timing out. Range is 1 to 20.

Defaults

3 seconds

Command Modes

Global configuration

Command History

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

Usage Guidelines

This command determines when the router abandons its attempt to connect to an RTSP server and declares a timeout error, if a connection cannot be established after the specified number of seconds.

Examples

The following example sets the connection timeout to 10 seconds:

```
rtsp client timeout connect 10
```

Related Commands

Command	Description
rtsp client session history records	Sets the maximum number of records to store in the RTSP client session history.
rtsp client timeout message	Sets the number of seconds that the router waits for a response from an RTSP server.

rtsp client timeout message

To set the number of seconds that the router waits for a response from a Real -Time Streaming Protocol (RTSP) server, use the **rtsp client timeout message** command in global configuration mode. To reset to the default, use the **no** form of this command.

rtsp client timeout message *seconds*

no rtsp client timeout message

Syntax Description

seconds	How long, in seconds, the router waits for a response from the server after making a request. Range is 1 to 20.

Defaults

3 seconds

Command Modes

Global configuration

Command History

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

Usage Guidelines

This command sets how long the router waits for the RTSP server to respond to a request before declaring a timeout error.

Examples

The following example sets the request timeout to 10 seconds:

```
rtsp client timeout message 10
```

Related Commands

Command	Description
rtsp client session history records	Sets the maximum number of records to store in the RTSP client session history.
rtsp client timeout connect	Sets the number of seconds allowed for the router to establish a TCP connection to an RTSP server.

rule

To apply a translation rule to a calling-party number or a called-party number for both incoming and outgoing calls, use the **rule** command in translation-rule configuration mode. To remove the translation rule, use the **no** form of this command.

rule *name-tag input-matched-pattern substituted-pattern* [*match-type substituted-type*]

no rule *name-tag input-matched-pattern substituted-pattern* [*match-type substituted-type*]

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.
input-matched-pattern	Input string of digits for which pattern matching is performed.
substituted-pattern	Replacement digit string that results after pattern matching is performed. Regular expressions are used to carry out this process.
match-type	(Optional) Choices for this field are **international**, **national**, **subscriber**, **abbreviated**, **unknown**, and **any**, as defined by the International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Q.931 specification. If you enter the *match-type* value, then you must also enter the *substituted-type* value.
substituted-type	(Optional) Choices for this field are **international**, **national**, **subscriber**, **abbreviated** and **unknown**, as defined by the ITU Q.931 specification.

Note In the syntax description above, the square brackets indicate optional values. When using this command, do not include these square brackets as part of the syntax. They are not valid parameters in the **rule** command. The square brackets can only be used in actual syntax for such commands as the **destination-pattern** and **incoming called-number** commands, where the syntax specifically allows this delimiter.

Defaults

No default behavior or values.

Command Modes

Translation-rule configuration

Command History

Release	Modification
12.0(7)XR1	This command was introduced on the Cisco AS5300.
12.0(7)XKs	This command was implemented on the following platforms: 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 was implemented on the following platforms: Cisco 1750, Cisco 2600 series, Cisco 3600 series, Cisco AS5300, Cisco 7200, and Cisco 7500.
12.1(2)T	This command was implemented on the Cisco MC3810.

Usage Guidelines

When configuring your dial peers, you are provided with an option called the *translation rule*. This option applies a translation rule to a calling party number (Automatic Number Identification [ANI]) or a called party number (Dial Number Information Service [DNIS]) for both incoming and outgoing calls within Cisco H.323 voice-enabled gateways. Also, the rule allows translation of the *type of number*.

Examples

The following example applies a translation rule. If a called number starts with 5552205 or 52205, then translation rule 21 uses the rule command to forward the number to 14085552205 instead.

```
translation-rule 21
 rule 1 555.% 1408555 subscriber international
 rule 2 7.% 1408555 abbreviated international
```

In the next example, if a called number is either 14085552205 or 014085552205, then after the execution of the translation rule 345 the forwarding digits are 52205. If the match type is configured and the type is not “unknown,” then dial-peer matching is required to match input string numbering type.

```
translation-rule 345
 rule 1 .%555.% 7 any abbreviated
```

Related Commands

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

rule (ENUM configuration)

To define a rule for an ENUM match table, use the **rule** command in ENUM configuration mode. To delete the rule, use the **no** form of this command.

rule *rule-number preference* {/*match-pattern*/ /*replacement-rule*/ *domain-name*}

no rule *rule-number preference* {/*match-pattern*/ /*replacement-rule*/ *domain-name*}

Syntax Description

rule-number	Assigns an identification number to the rule. Range is from 1 to 2147483647.
preference	Assigns a preference value to the rule. Range is from 1 to 2147483647. Lower values have higher preference.
match-pattern	Stream editor (SED) expression used to match incoming call information. The slash "/" is a delimiter in the pattern.
replacement-rule	SED expression used to replace match-pattern in the call information. The slash "/" is a delimiter in the pattern.
domain-name	Domain name to be used while the query to the DNS server is sent.

Defaults

No default behavior or values

Command Modes

ENUM configuration

Command History

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

Usage Guidelines

The following table shows examples of match patterns, input strings, and result strings.

Match Pattern	Replacement Pattern	Input String	Result String	Description
`/^$/`	`//`			Null string to null string.
`/^.*/`	`//`	`4085552711`		Any string to null string.
`/^456\(.*\)/`	`/555\1/`	`5557123`	`5557123`	Match from the beginning of the input string.
`/\(^...\)456\(...\)/`	`/\1555\2/`	`408555777`	`4085557777`	Match string from the middle of the input string.
`/\(.*\)8920/`	`/\15555/`	`4085558920`	`4081115555`	Match from the end of the input string.
`/^1#\(.*\)/`	`/\1/`	`1#2345`	`2345`	Replace match string with null string.
`/^408...\(8333\)/`	`/555\1/`	`4085558333`	`5558333`	Match multiple patterns.

Rules are entered in any order, but their preference number determines the sequence in which they are used for matching against the input string, which is a called number. A lower preference number is used before a higher preference number.

If a match is found, the input string is modified according to the replacement rule, and the E.164 domain name is attached to the modified number. This longer number is sent to a Domain Name System (DNS) server to determine a destination for the call. The server returns one or more URLs as possible destinations. The originating gateway tries to place the call using each URL in order of preference. If a call cannot be completed using any of the URLs, the call is disconnected.

Examples

The following example defines ENUM rule number 3 with preference 2. The beginning of the call string is checked for digits 9011; when a match is found, 9011 is replaced with 1408 and the call is sent out as an e164.arpa number.

```
Router(config)# voice enum-match-table number
Router(config-enum)# rule 3 2 /^9011\(.*\)//+1408\1/arpa
```

Related Commands

Command	Description
show voice enum-match-table	Displays the configuration of a voice ENUM match table.
test enum	Tests the ENUM rule.
voice enum-match-table	Initiates the definition of a voice ENUM match table.

rule (voice translation-rule)

To define a translation rule, use the **rule** command in voice translation-rule configuration mode. To delete the translation rule, use the **no** form of this command.

Match and Replace Rule

rule *precedence /match-pattern/ /replace-pattern/*
[**type** {*match-type replace-type*} [**plan** {*match-type replace-type*}]]

no rule *precedence*

Reject Rule

rule *precedence* **reject** */match-pattern/* [**type** *match-type* [**plan** *match-type*]]

no rule *precedence*

Syntax Description

precedence	Priority of the translation rule. Range is from 1 to 15.
/match-pattern/	Stream editor (SED) expression used to match incoming call information. The slash '/' is a delimiter in the pattern.
/replace-pattern/	SED expression used to replace the match pattern in the call information. The slash '/' is a delimiter in the pattern.
type *match-type replace-type*	(Optional) Number type of the call. Valid values for the *match-type* argument are as follows: • **abbreviated**—Abbreviated representation of the complete number as supported by this network. • **any**—Any type of called number. • **international**—Number called to reach a subscriber in another country. • **national**—Number called to reach a subscriber in the same country, but outside the local network. • **network**—Administrative or service number specific to the serving network. • **reserved**—Reserved for extension.**subscriber**—Number called to reach a subscriber in the same local network. • **unknown**—Number of a type that is unknown by the network. Valid values for the *replace-type* argument are as follows: • **abbreviated**—Abbreviated representation of the complete number as supported by this network. • **international**—Number called to reach a subscriber in another country. • **national**—Number called to reach a subscriber in the same country, but outside the local network.

type *match-type replace-type* (continued)	• **network**—Administrative or service number specific to the serving network. • **reserved**—Reserved for extension. • **subscriber**—Number called to reach a subscriber in the same local network. • **unknown**—Number of a type that is unknown by the network.
plan *match-type replace-type*	(Optional) Numbering plan of the call. Valid values for the *match-type* argument are as follows: • **any**—Any type of dialed number. • **data** • **ermes** • **isdn** • **national**—Number called to reach a subscriber in the same country, but outside the local network. • **private** • **reserved**—Reserved for extension. • **telex** • **unknown**—Number of a type that is unknown by the network. Valid values for the *replace-type* argument are as follows: • **data** • **ermes** • **isdn** • **national**—Number called to reach a subscriber in the same country, but outside the local network. • **private** • **reserved**—Reserved for extension. • **telex** • **unknown**—Number of a type that is unknown by the network.
reject	The match pattern of a translation rule is used for call-reject purposes.

Defaults

No default behavior or values

Command Modes

Voice translation-rule configuration

Command History

Release	Modification
12.2(11)T	This command was introduced with a new syntax in voice-translation-rule configuration mode.

Usage Guidelines

> **Note** The **rule** command introduced in this feature is used under the **voice translation-rule** command. An earlier version of this command uses the same name but is used under the **translation-rule** command and has a slightly different command syntax. In the older version, you cannot use the square brackets when you are entering command syntax. They appear in the syntax only to indicate optional parameters, but are not accepted as delimiters in actual command entries. In the newer version, you can use the square brackets as delimiters. Going forward, Cisco recommends that you use this newer version to define rules for call matching. Eventually, the **translation-rule** command will not be supported.

A translation rule applies to a calling party number (automatic number identification [ANI]) or a called party number (dialed number identification service [DNIS]) for incoming, outgoing, and redirected calls within Cisco H.323 voice-enabled gateways.

Number translation occurs several times during the call routing process. In both the originating and terminating gateways, the incoming call is translated before an inbound dial peer is matched, before an outbound dial peer is matched, and before a call request is set up. Your dial plan should account for these translation steps when translation rules are defined.

Each rule consists of SED-like expressions for the matching and replacement patterns, and may include any of the following components:

- Escape sequences using back slashes
- Keywords “NULL” and “ANY”
- A CTRL-v before a question mark (“?”) in order to use the question mark as a symbol in a match pattern
- Either “&” or “\0” for copying the substring matched by the match pattern

The following table shows examples of match patterns, input strings, and result strings.

Match Pattern	Replacement Pattern	Input String	Result String	Description
`/^$/`	`//`			Null string to null string.
`/^.*/`	`//`	`4085662711`		Any string to null string.
`//`	`//`	`4085551234`	`4085551234`	Match any string but no replacement. Use this to manipulate the call plan or call type.
`/^456\(.*\)/`	`/555\1/`	`4567123`	`5557123`	Match from the beginning of the input string.
`/\(^...\)456\(...\)/`	`/\1555\2/`	`4084567777`	`4085557777`	Match from the middle of the input string.
`/\(.*\)8920/`	`/\15555/`	`4081118920`	`4081115555`	Match from the end of the input string.
`/^1#\(.*\)/`	`/\1/`	`1#2345`	`2345`	Replace match string with null string.
`/^408...\(8333\)/`	`/555\1/`	`4087778333`	`5558333`	Match multiple patterns.
`/1234/`	`/00&00/`	`5551234`	`55500123400`	Match the substring.
`/1234/`	`/00\000/`	`5551234`	`55500123400`	Match the substring (same as &).

The software verifies that a replacement pattern is in a valid E.164 format that can include the permitted special characters. If the format is not valid, the expression is treated as an unrecognized command.

The number type and calling plan are optional parameters for matching a call. If either parameter is defined, the call is checked against the match pattern and the selected type or plan value. If the call matches all the conditions, the call is accepted for additional processing, such as number translation.

Several rules may be grouped together into a translation rule, which gives a name to the rule set. A translation rule may contain up to 15 rules. All calls that refer to this translation rule are translated against this set of criteria.

The precedence value of each rule may be used in a different order than that in which they were typed into the set. Each rule's precedence value specifies the priority order in which the rules are to be used. For example, rule 3 may be entered before rule 1, but the software uses rule 1 before rule 3.

The software supports up to 128 translation rules. A translation profile collects and identifies a set of these translation rules for translating called, calling, and redirected numbers. A translation profile is referenced by trunk groups, source IP groups, voice ports, dial peers, and interfaces for handling call translation.

Examples

The following example applies a translation rule. If a called number starts with 5552205 or 72205, translation rule 21 uses the rule command to forward the number to 14085552205 instead.

```
Router(config)# voice translation-rule 21
Router(cfg-translation-rule)# rule 1 /^5552205/ /14085552205/
Router(cfg-translation-rule)# rule 2 /^72205/ /14085552205/
```

In the next example, if a called number is either 14085552205 or 014085552205, after the execution of translation rule 345, the forwarding digits are 52205. If the match type is configured and the type is not "unknown," dial-peer matching is required to match the input string numbering type.

```
Router(config)# voice translation-rule 345
Router(cfg-translation-rule)# rule 1 /^14085552205/ /52205/ plan any national
Router(cfg-translation-rule)# rule 2 /^014085552205/ /52205/ plan any national
```

Related Commands

Command	Description
show voice translation-rule	Displays the parameters of a translation rule.
voice translation-rule	Initiates the voice translation-rule definition.