



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 Library*.

radius-server attribute 6

To provide for the presence of the Service-Type attribute (attribute 6) in RADIUS Access-Accept messages, use the **radius-server attribute 6** command in global configuration mode. To make the presence of the Service-Type attribute optional in Access-Accept messages, use the **no** form of this command.

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

no radius-server attribute 6 { **mandatory** | **on-for-login-auth** | **support-multiple** | **voice** *value* }

Syntax Description

mandatory	Makes the presence of the Service-Type attribute mandatory in RADIUS Access-Accept messages.
on-for-login-auth	Sends the Service-Type attribute in the authentication packets. Note The Service-Type attribute is sent by default in RADIUS Accept-Request messages. Therefore, RADIUS tunnel profiles should include “Service-Type=Outbound” as a check item, not just as a reply item. Failure to include Service-Type=Outbound as a check item can result in a security hole.
support-multiple	Supports multiple Service-Type values for each RADIUS profile.
voice <i>value</i>	Selects the Service-Type value for voice calls. The only value that can be entered is 1. The default is 12.

Command Default

If this command is not configured, the absence of the Service-Type attribute is ignored, and the authentication or authorization does not fail. The default for the **voice** keyword is 12.

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	This command was introduced.
12.2(13)T	The mandatory keyword was added.

Usage Guidelines

If this command is configured and the Service-Type attribute is absent in the Access-Accept message packets, the authentication or authorization fails.

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 an Access-Accept packet containing multiple instances being treated as though an Access-Reject was received.

Examples

The following example shows that the presence of the Service-Type attribute is mandatory in RADIUS Access-Accept messages:

```
Router(config)# radius-server attribute 6 mandatory
```

The following example shows that attribute 6 is to be sent in authentication packets:

```
Router(config)# radius-server attribute 6 on-for-login-auth
```

The following example shows that multiple Service-Type values are to be supported for each RADIUS profile:

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

The following example shows that Service-Type values are to be sent in voice calls:

```
Router(config)# radius-server attribute 6 voice 1
```

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.
<i>value</i>	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.

Command Default

```
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 RAS message timeout values.

ras retry lrq

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

ras retry lrq *value*

no ras retry lrq

Syntax Description

lrq	Configures the location request (LRQ) message counter.
<i>value</i>	Number of times for the zone gatekeeper (ZGK) to resend messages to the directory gatekeeper (DGK) after the timeout period. The timeout period is the period in which a message has not been received by the ZGK from the DGK and is configured using the ras timeout lrq command. Valid values are 1 through 30.

Command Default

The retry counter is set to 1.

Command Modes

Gatekeeper configuration

Command History

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

Usage Guidelines

Use this command in conjunction with the **ras timeout lrq** command. The **ras timeout lrq** command configures the number of seconds for the gateway to wait before resending a RAS message to a gatekeeper. The **ras retry lrq** 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 LRQ message counter set to 5:

```
Router(conf-gk) # ras retry lrq 5
```

Related Commands

Command	Description
ras timeout lrq	Configures the gatekeeper 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.

Command Default

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.
12.3(4)T	The default change implemented in Cisco IOS Release 12.3(3) was integrated in Cisco IOS Release 12.3(4)T.

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 seconds [margin seconds]
```

```
no ras rrq ttl
```

Syntax Description

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

Command Default

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

Command Modes

Voice service h323 configuration

Command History

Release	Modification
12.3(1)	This command was introduced.
12.3(6)	The maximum <i>time-to-live</i> value was changed from 300 to 4000 seconds.
12.3(4)T2	The maximum <i>time-to-live</i> value was changed from 300 to 4000 seconds.
12.3(7)T	The maximum <i>time-to-live</i> 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 seconds* value configured to 300 seconds and the **margin seconds** 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} seconds
```

```
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.
<i>seconds</i>	Number of seconds for the gateway to wait for a message from the gatekeeper before timing out. Valid values are 1 through 45.

Command Default

```
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 RAS message retry counters.

ras timeout decisec

To configure the H.323 Registration, Admission, and Status (RAS) message timeout values in deciseconds, use the **ras timeout decisec** 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} decisec decisecond
```

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

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. Default: 3.
brq	Configures the bandwidth request (BRQ) message timer. Default: 3.
drq	Configures the disengage request (DRQ) message timer. Default: 3.
grq	Configures the gatekeeper request (GRQ) message timer. Default: 5.
rai	Configures the resource availability indication (RAI) message timer. Default: 3.
rrq	Configures the registration request (RRQ) message timer. Default: 5.
<i>decisecond</i>	Number of deciseconds for the gateway to wait for a message from the gatekeeper before timing out. Valid values are 1 through 45.

Command Default

Timers are set to their default values.

Command Modes

Voice service h323 configuration

Command History

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

Usage Guidelines

Use this command in conjunction with the **ras retry** command. The **ras timeout decisec** command configures the number of deciseconds 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 ARQ message timeout value set to 25 deciseconds and all other RAS message timeout values set to 30 deciseconds:

```
Router(conf-serv-h323)# ras timeout arq decisec 25  
Router(conf-serv-h323)# ras timeout all decisec 30
```

Related Commands

Command	Description
ras retry	Configures the H.323 RAS message retry counters.
ras timeout	Configures the H.323 RAS message timeout values in seconds.

ras timeout lrq

To configure the Gatekeeper Registration, Admission, and Status (RAS) message timeout values, use the **ras timeout lrq** command in gatekeeper configuration mode. To set the timers to the default values, use the **no** form of this command.

ras timeout lrq *seconds*

no ras timeout lrq

Syntax Description	lrq	Configures the location request (LRQ) message timer.
	<i>seconds</i>	Number of seconds for the zone gatekeeper (ZGK) to wait for a message from the directory gatekeeper (DGK) before timing out. Valid values are 1 through 45. The default is 2.

Command Default Timers are set to their default value

Command Modes Gatekeeper configuration

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

Usage Guidelines Use this command in conjunction with the **ras retry lrq** command. The **ras timeout lrq** command configures the number of seconds for the zone gatekeeper (ZGK) to wait before resending a RAS message to a directory gatekeeper (DGK). The **ras retry lrq** 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 gatekeepers. For example, if you have gatekeepers that are slow to respond to a LRQ 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 LRQ message timeout value set to 4 seconds:

```
Router(conf-gk)# ras timeout lrq 4
```

Related Commands	Command	Description
	ras retry lrq	Configures the gatekeeper 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 1AESS 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.

Command Default

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 supports the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5800.

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 1AESS 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.

reason-header override

To enable cause code passing from one SIP leg to another, use the **reason-header override** command in SIP UA configuration mode. To disable reason-header override, use the **no** form of this command.

reason-header override

no reason-header override

Syntax Description This command has no arguments or keywords.

Command Default No default behavior or values.

Command Modes SIP UA configuration

Command History	Release	Modification
	12.3(8)T	This command was introduced.
	12.4(9)T	Usage guidelines were updated to include configuration requirements for SIP-to-SIP configurations.

Usage Guidelines In an SIP-to-SIP configuration the **reason-header override** command must be configured to ensure cause code passing from the incoming SIP leg to the outgoing SIP leg.

Examples The following example, shows the SIP user agent with reason-header override being configured.

```
Router(config)# sip-ua
Router(config-sip-ua)# reason-header override
```

Related Commands	Command	Description
	sip-ua	Enables SIP UA configuration commands.

redial

To designate an SCCP telephony control (STC) application feature speed-dial code to redial the last number dialed, use the **redial** command in STC application feature speed-dial configuration mode. To return the code to its default, use the **no** form of this command.

redial *keypad-character*

no redial

Syntax Description

<i>keypad-character</i>	Single character that can be dialed on a telephone keypad. Default is # (pound sign).
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Command Default

The default code for redial is # (pound sign).

Command Modes

STC application feature speed-dial configuration

Command History

Release	Modification
12.4(2)T	This command was introduced.

Usage Guidelines

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

Use this command to set the redial code only if you want to change the value from its default, which is # (pound sign).

To use the redial feature on a phone, dial the STC application feature speed-dial (FSD) prefix and the code that has been configured with this command (or the default if this command was not used). For example, if the FSD prefix is * (the default), and you want to redial the phone number most recently dialed from this line, dial *#.

If a call has never been made from a phone, and you dial the redial prefix and code, you hear silence because the call-control system does not have a number to redial and cannot respond.

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

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

Examples

The following example sets an FSD code prefix of two pound signs (##) and a redial code of 9. After these values are configured, a phone user presses ##9 on the keypad to redial the last number called from the phone.

```
Router(config)# stcapp feature speed-dial
Router(stcapp-fsd)# prefix ##
Router(stcapp-fsd)# redial 9
Router(stcapp-fsd)# exit
```

Related Commands

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

redirect contact order

To set the order of contacts in the 300 Multiple Choice message, use the **redirect contact order** command in SIP configuration mode. To reset the order of contacts to the default, use the **no** form of this command.

redirect contact order [best-match | longest-match]

no redirect contact order

Syntax Description	best-match	(Optional) Uses the current system configuration.
	longest-match	(Optional) Uses the destination pattern longest match first, and then the second longest match, the third longest match, and so on. This is the default.

Command Default longest-match

Command Modes SIP configuration

Command History	Release	Modification
	12.2(15)ZJ	This command was introduced.
	12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines This command applies when a 300 Multiple Choice message is sent by a SIP gateway indicating that a call has been redirected and that there are multiple routes to the destination.

Enter SIP configuration mode after entering voice service VoIP configuration mode as shown in the following example.

Examples The following example uses the current system configuration to set the order of contact:

```
Router(config)# voice service voip
Router(config-voi-srv)# sip
Router(conf-serv-sip)# redirect contact order best-match
```

Related Commands	Command	Description
	sip	Enters SIP configuration mode.

redirect ip2ip (dial peer)

To redirect SIP phone calls to SIP phone calls on a specific VoIP dial peer using the Cisco IOS Voice Gateway, use the **redirect ip2ip** command in dial peer configuration mode. To disable redirection, use the **no** form of this command.

redirect ip2ip

no redirect ip2ip

Syntax Description This command has no arguments or keywords.

Command Default Redirection is disabled.

Command Modes Dial peer configuration

Command History	Release	Modification
	12.2(15)ZJ	This command was introduced.
	12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines The **redirect ip2ip** command must be configured on the inbound dial peer of the gateway. This command enables, on a per dial peer basis, IP-to-IP call redirection for the gateway.

To enable global IP-to-IP call redirection for all VoIP dial peers, use voice service configuration mode. To specify IP-to-IP call redirection for a specific VoIP dial peer, configure the dial peer in dial peer configuration mode.



Note

When IP-to-IP redirection is configured in dial peer configuration mode, the configuration for the specific dial peer is activated only if the dial peer is an inbound dial peer. To enable IP-to-IP redirection globally, use **redirect ip2ip** (voice service) command.

Examples The following example specifies that on VoIP dial peer 99, IP-to-IP redirection is set:

```
dial-peer voice 99 voip
  redirect ip2ip
```

Related Commands	Command	Description
	redirect ip2ip (voice service)	Redirects SIP phone calls to SIP phone calls globally on a gateway using the Cisco IOS voice gateway.

redirect ip2ip (voice service)

To redirect SIP phone calls to SIP phone calls globally on a gateway using the Cisco IOS Voice Gateway, use the **redirect ip2ip** command in voice service configuration mode. To disable redirection, use the **no** form of this command.

redirect ip2ip

no redirect ip2ip

Syntax Description This command has no arguments or keywords.

Command Default Redirection is disabled.

Command Modes Voice service configuration

Command History	Release	Modification
	12.2(15)ZJ	This command was introduced.
	12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines Use this command to enable IP-to-IP call redirection globally on a gateway. Use the **redirect ip2ip** (dial peer) command to configure IP-to-IP redirection on a specific inbound dial peer.

Examples The following example specifies that all VoIP dial peers use IP-to-IP redirection:

```
voice service voip
  redirect ip2ip
```

Related Commands	Command	Description
	redirect ip2ip (dial peer)	Redirects SIP phone calls to SIP phone calls on a specific VoIP dial peer using the Cisco IOS voice gateway.

redirection (SIP)

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

redirection

no redirection

Syntax Description This command has no arguments or keywords.

Command Default Redirection is enabled.

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

Examples The following example disables processing of incoming 3xx 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.

refer-ood enable

To enable out-of-dialog refer (OOD-R) processing, use the **refer-ood enable** command in SIP user-agent configuration mode. To disable OOD-R, use the **no** form of this command.

refer-ood enable [*request-limit*]

no refer-ood enable

Syntax Description	<i>request-limit</i>	(Optional) Maximum number of concurrent incoming OOD-R requests that the router can process. Range: 1 to 500. Default: 500.
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Command Default	OOD-R processing is disabled.	
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Command Modes	SIP UA configuration (config-sip-ua)	
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Command History	Release	Modification
	12.4(11)XJ	This command was introduced.
	12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines	Out of dialog Refer allows applications to establish calls using the SIP gateway or Cisco Unified CME. The application sets up the call and the user does not dial out from their own phone.
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Examples	The following example shows how to enable OOD-R:
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```
Router(config)# sip-ua
Router(config-sip-ua)# refer-ood enable
```

Related Commands	Command	Description
	authenticate (voice register global)	Defines the authenticate mode for SIP phones in a Cisco Unified CME or Cisco Unified SRST system.
	credential load	Reloads a credential file into flash memory.
	debug voip application	Displays all application debug messages.

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.

Command Default 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, +5550112, 5550112, and 4085550112 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 5550112
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-pattern 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

<i>cadence</i>	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.
----------------	--

Command Default

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
```

Related Commands

Command	Description
destination-pattern not-provided	Specifies the port to receive the incoming calls that have no called-party number.

registrar

To enable Session Initiation Protocol (SIP) gateways to register E.164 numbers on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and SCCP phones with an external SIP proxy or SIP registrar, use the **registrar** command in SIP UA configuration mode. To disable registration of E.164 numbers, use the **no** form of this command.

```
registrar { dns:address | ipv4:destination-address } aor-domain expires seconds [tcp [tls]] type
[secondary] [scheme string]
```

```
no registrar [secondary]
```

Syntax	Description
dns:address	DNS address of the primary SIP registrar server.
ipv4:destination-address	IP address of the primary SIP registrar server.
aor-domain	Use address-of-record (AOR) domain name in To/From headers of the outgoing register.
expires <i>seconds</i>	Default registration time, in seconds. Range is 60 to 65535. Default is 3600.
tcp	TCP transport layer protocol. When the tcp keyword is not selected, UDP is the default.
tls	(Optional) TLS over TCP transport layer protocol. When the tcp tls keyword is not selected, UDP is the default.
type	Registration type.
secondary	(Optional) Registration with a secondary SIP proxy or registrar to provide redundancy if the primary registrar fails.
scheme <i>string</i>	URL scheme. String set to either <i>sip</i> or <i>sips</i> . Default is <i>sip</i> .

Command Default Registration is disabled.

Command Modes SIP UA configuration

Command History	Release	Modification
	12.2(15)ZJ	This command was introduced.
	12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.
	12.4(6)T	Added the following optional tls keyword and keyword and argument: scheme <i>string</i> .

Usage Guidelines This command enables the gateway to register E.164 telephone numbers with primary and secondary external SIP registrars.

By default, SIP gateways do not generate SIP register messages.

Examples

The following example specifies registration with a primary and secondary registrar server.

```
.
.
.
sip-ua
  retry invite 3
  retry register 3
  timers register 150
  registrar ipv4:10.8.17.40 expires 3600 secondary
.
.
.
```

The following example specifies an IP address with TCP.

```
.
.
.
sip-ua
  retry invite 3
  retry register 3
  timers register 150
  registrar ipv4:10.8.17.40 tcp
.
.
.
```

The following example specifies an IP address with TLS over TCP.

```
.
.
.
sip-ua
  retry invite 3
  retry register 3
  timers register 150
  registrar ipv4:10.8.17.40 tcp tls
.
.
.
```

The following example specifies a URL scheme with SIP security (SIPS).

```
.
.
.
sip-ua
  retry invite 3
  retry register 3
  timers register 150
  registrar ipv4:10.8.17.40 scheme sips
.
.
.
```

Related Commands

Command	Description
retry register	Sets the total number of SIP Register messages to send.
show sip-ua register status	Displays the status of E.164 numbers that a SIP gateway has registered with an external primary or secondary SIP registrar.
timers register	Sets how long the SIP UA waits before sending register requests.

registration retries

To set the number of times that Skinny Client Control Protocol (SCCP) tries to register with a Cisco Unified CallManager, use the **registration retries** command in SCCP Cisco CallManager configuration mode. To reset this number to the default value, use the **no** form of this command.

registration retries *retry-attempts*

no registration retries

Syntax Description	<i>retry-attempts</i>	Number of registration attempts. Range is 1 to 32. Default is 3.
---------------------------	-----------------------	--

Command Default	3 registration attempts
------------------------	-------------------------

Command Modes	SCCP Cisco CallManager configuration
----------------------	--------------------------------------

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

Usage Guidelines	Use this command to control the number of registration retries before SCCP confirms that it cannot register with the Cisco Unified CallManager. When SCCP confirms that it cannot register to the current Cisco Unified CallManager (if the number of registration requests sent without an Ack reaches the registration retries value), SCCP tries to register with the next Cisco Unified CallManager.
-------------------------	--



Note

The optimum setting for this command depends on the platform and your individual network characteristics. Adjust the registration retry attempts to meet your needs.

Examples	The following example sets the number of registration retries to 15:
-----------------	--

```
Router(config-sccp-cm) # registration retries 15
```

Related Commands	Command	Description
	ccm group	Creates a Cisco Unified CallManger group and enters SCCP Cisco CallManager configuration mode.
	registration timeout	Sets the length of time between registration messages sent from SCCP to the Cisco CallManager.

registration timeout

To set the length of time between registration messages sent from Skinny Client Control Protocol (SCCP) to the Cisco Unified CallManager, use the **registration timeout** command in SCCP Cisco CallManager configuration mode. To reset the length of time to the default value, use the **no** form of this command.

registration timeout *seconds*

no registration timeout

Syntax Description	<i>seconds</i>	Time, in seconds, between registration messages. Range is 1 to 180. Default is 3.
---------------------------	----------------	---

Command Default	3 seconds
------------------------	-----------

Command Modes	SCCP Cisco CallManager configuration
----------------------	--------------------------------------

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

Usage Guidelines	Whenever SCCP sends the registration message to the Cisco Unified CallManager, it initiates this timer. Once the timeout occurs, it sends the next registration message unless the number of messages without an Ack reaches the number set by the registration retries command. Use this command to set the Cisco Unified CallManager registration timeout parameter value.
-------------------------	---



Note

The optimum setting for this command depends on the platform and your individual network characteristics. Adjust the registration timeout value to meet your needs.

Examples	The following example sets the length of time between registration messages sent from SCCP to the Cisco Unified CallManager to 12 seconds:
-----------------	--

```
Router(config-sccp-cm) # registration timeout 12
```

Related Commands	Command	Description
	ccm group	Creates a Cisco CallManger group and enters SCCP Cisco CallManager configuration mode.
	registration retries	Sets the number of times that SCCP tries to register with the Cisco Unified CallManager.

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 <i>value</i>	Supports reliable provisional responses. The <i>value</i> 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 <i>value</i> of 100rel, is the default.
require <i>value</i>	Requires reliable provisional responses. The <i>value</i> 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.

Command Default

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 supported on 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 configuration mode. You can configure reliable provisional responses for the specific dial peer only by using the **voice-class sip rel1xx** command.
- SIP configuration 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 UA configuration mode. To disable Remote-Party-ID translation, use the **no** form of this command.

remote-party-id

no remote-party-id

Syntax Description This command has no arguments or keywords.

Command Default Remote-Party-ID translation is enabled

Command Modes SIP UA 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 restore the default value for this command, use the **no** form of this command.

```
req-qos {best-effort | controlled-load | guaranteed-delay} [{audio bandwidth | video
bandwidth} default | max bandwidth-value]
```

```
no req-qos
```

Syntax Description		
best-effort		Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation.
controlled-load		Indicates that 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		Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.
audio bandwidth		(Optional) Specifies amount of bandwidth to be requested for audio streams.
default		Sets the default bandwidth to be requested for audio or video streams. <ul style="list-style-type: none"> • Audio streams—Range is 1 to 64 kbps; default value is 64 kbps. • Video streams—Range is 1 to 5000 kbps; default value is no maximum
max <i>bandwidth-value</i>		Sets the maximum bandwidth to be requested for audio streams. Range is 1 to 64 kbps; default value is no maximum.
video bandwidth		(Optional) Specifies the amount of bandwidth to be requested for video streams.
default <i>bandwidth-value</i>		Sets the default bandwidth to be requested for video streams. Range is 1 to 5000 kbps; default value is 384 kbps.
max <i>bandwidth-value</i>		(Optional) Sets the maximum bandwidth to be requested for video streams.

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 routers.
12.3(4)T	Keywords added to support audio and video streams.

Usage Guidelines

Use the **req-qos** 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.

This command is applicable only to VoIP dial peers.

Examples

The following example configures guaranteed-delay as the requested quality of service to a dial peer:

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

The following example configures guaranteed-delay and requests a default bandwidth level of 768 kbps for video streams:

```
dial-peer voice 20 voip
  req-qos guaranteed-delay video bandwidth default 768
```

Related Commands

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

request (XML transport)

To set the XML transport mode request handling parameters, use the **request** command in XML transport configuration mode. To disable the XML transport request parameter setting, use the **no** form of this command

```
request { outstanding number | timeout seconds }
```

```
no request
```

Syntax Description

outstanding	Maximum number of outstanding requests.
<i>number</i>	The valid range for the number of outstanding requests is from 1 to 10. The default is 1.
timeout	Response timeout at the transport level.
<i>seconds</i>	Specifies the number of seconds a request is active before it times out. Valid range is from 0 to 60 seconds. The default value is 0 (no timeout).

Command Default

The default for **outstanding** is 1 and the default for **timeout** is 0 (no timeout).

Command Modes

XML transport configuration

Command History

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

Usage Guidelines

Use this command to set the request timeout. A value of 0 seconds specifies no timeout. This timeout applies to the request being processed and not outstanding requests as described below. The specified timeout limits the amount of time between the request being dequeued by the application and the completion of the processing of that request.

Use this command to specify the number of outstanding requests allowed per application for the specified transport mode. The outstanding requests are those requests that are queued at the application for processing but have not yet been processed.

Examples

The following example shows how to enter XML transport configuration mode, set the XML transport request timeout to 10 seconds, and exit XML transport configuration mode:

```
Router(config)# ixi transport http
Router(conf-xml-trans)# request timeout 10
```

Related Commands	Command	Description
	ixi transport http	Enters XML transport configuration mode.
	ixi application mib	Enters XML application configuration mode.
	response size (XML transport)	Set the XML transport fragment size.

reset

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

reset *number*

Syntax Description	<i>number</i>	Number of DSPs to be reset. Range is from 0 to 30.
Command Default	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
```

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**]

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.

Command Default

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.
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.

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.

resource-pool (mediacard)

To create a Digital Signal Processor (DSP) resource pool on ad-hoc conferencing and transcoding port adapters, use the **resource-pool** command in mediacard configuration mode. To remove the DSP resource pool and release the associated DSP resources, use the **no** form of this command.

resource-pool *identifier* **dsps** *number*

no resource-pool *identifier* **dsps** *number*

Syntax Description	Parameter	Description
	<i>identifier</i>	Identifies the DSP resource to be configured. Valid values consist of alphanumeric characters, plus “_” and “-”.
	dsps	Digital signal processor.
	<i>number</i>	Specifies the number of DSPs to be allocated for the specified resource pool. Valid values are from 1 to 4.

Command Default No default behavior or values

Command Modes Mediacard configuration

Command History	Release	Modification
	12.3(8)XY	This command was introduced on the Communication Media Module.
	12.3(14)T	This command was integrated into Cisco IOS Release 12.3(14)T.
	12.4(3)	This command was integrated into Cisco IOS Release 12.4(3).

Usage Guidelines The DSP resource pool identifier should be unique across the same Communication Media Module (CMM). Removing a resource pool may cause the profile using that resource pool to be disabled if it is the last resource pool in the profile.

Examples The following example shows how to create a DSP resource pool:

```
resource-pool headquarters_location1 dsps 2
```

Related Commands	Command	Description
	debug mediacard	Displays debugging information for DSPRM.
	show mediacard	Displays information about the selected media card.

response (XML application)

To set XML application response parameters, use the **response** command in XML application configuration mode. To disable response parameter settings, use the **no** form of this command.

```
response { formatted | timeout { -1 | seconds } }
```

```
no response { formatted | timeout { -1 | seconds } }
```

Syntax Description	Parameter	Description
	formatted	Response parameters in formatted human readable XML.
	timeout	Application specified response timeout.
	-1	Enter -1 to indicate no application specified timeout. This is the default timeout setting.
	<i>seconds</i>	Number of seconds a response is active before it times out. Valid range includes 0 to 60 seconds.

Command Default The default for the **timeout** keyword is **-1** indicating not application specified timeout.

Command Modes XML application configuration

Command History	Release	Modification
	12.4(6)T	This command was introduced.

Usage Guidelines The response timeout specified in this command, if other than -1 which is the default, overwrites the timeout value specified in the request (XML transport) command that sets the timeout at the transport level.

The same http transport layer could have multiple applications active at the same time. You can set the timeout for each application individually or have all of the applications to use the same timeout value set at transport layer using the request (XML transport) command in XML transport configuration mode.

Examples The following example shows how to enter XML application configuration mode, set XML response parameters in formatted human readable XML, and exit XML application configuration mode:

```
Router(config)# ixi application mib
Router(conf-xml-app)# response formatted
```

Related Commands	Command	Description
	ixi application mib	Enters XML application configuration mode.
	request (XML transport)	Set the XML transport mode request handling parameters.

response size (XML transport)

To set the response transport fragment size, use the **response size** command in XML transport configuration mode. To disable the response transport fragment size setting, use the **no** form of this command.

response size *kBps*

no response size

Syntax Description	<i>kBps</i>	Size of the fragment in the response buffer in kilobytes. Valid range is 1 to 64 kB. The default is 4 kB.
---------------------------	-------------	---

Command Modes	XML transport configuration
----------------------	-----------------------------

Command History	Release	Modification
	12.4(6)T	This command was introduced.

Usage Guidelines	The fragment size is constrained by the transport type. The CLI help provides input guidelines.
-------------------------	---

Examples	The following example shows how to enter XML transport configuration mode, set XML transport fragment size to 32 Kbytes, and exit XML transport configuration mode:
-----------------	---

```
Router(config)# ixi transport http
Router(conf-xml-trans)# response size 32
```

Related Commands	Command	Description
	ixi transport http	Enters XML transport configuration mode.
	ixi application mib	Enter XML application configuration mode.
	request (XML transport)	Sets XML transport request handling parameters.

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 *seconds*

no response-timeout *seconds*

Syntax Description	<i>seconds</i>	Response waiting time, in seconds. Default is 1.
---------------------------	----------------	--

Command Default	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.

retries (auto-config application)

To set the number of download retry attempts for an auto-configuration application, use the **retries** command in auto-config application configuration mode. To reset to the default, use the **no** form of this command.

retries *number*

no retries

Syntax Description	<i>number</i>	Specifies the download retry attempts. Valid range is 1 to 3.
--------------------	---------------	---

Command Default	The default value is 2.
-----------------	-------------------------

Command Modes	Auto-config application configuration
---------------	---------------------------------------

Command History	Release	Modification
	12.3(8)XY	This command was introduced on the Communication Media Module.
12.3(14)T	This command was integrated into Cisco IOS Release 12.3(14)T.	

Examples	The following example shows the retries command used to set the number of retries for an auto-configuration application to 3:
----------	--

```
Router(auto-config-app)# retries 3
```

Related Commands	Command	Description
	auto-config	Enables auto-configuration or enters auto-config application configuration mode for the SCCP application.
show auto-config	Displays the current status of auto-configuration applications.	

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 UA configuration mode. To reset to the default, use the **no** form of this command.

retry bye *number*

no retry bye *number*

Syntax Description	<i>number</i>	Number of BYE retries. Range is from 1 to 10. The default is 10.
---------------------------	---------------	--

Command Default	10 retries
------------------------	------------

Command Modes	SIP UA 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 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 UA configuration mode. To reset to the default, use the **no** form of this command.

retry cancel *number*

no retry cancel *number*

Syntax Description	<i>number</i>	Number of CANCEL retries. Range is from 1 to 10. Default is 10.
--------------------	---------------	---

Command Default	10 retries
-----------------	------------

Command Modes	SIP UA 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 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 UA configuration mode. To reset to the default, use the **no** form of this command.

retry comet *number*

no retry comet

Syntax Description	<i>number</i>	Number of COMET retries. Range is from 1 to 10. Default is 10.
---------------------------	---------------	--

Command Default	10 retries
------------------------	------------

Command Modes	SIP UA 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.

Command	Description
retry invite	Configures the number of times that a 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 usage configuration mode. To reset to the default, use the **no** form of this command.

retry interval *seconds*

no retry interval

Syntax Description	<i>seconds</i>	Retry interval between delivery attempts, in seconds. Range is from 1 to 3600 (1 hour). The default is 900.
---------------------------	----------------	---

Command Default	900 seconds
------------------------	-------------

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 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 UA configuration mode. To reset to the default, use the **no** form of this command.

retry invite *number*

no retry invite *number*

Syntax Description	<i>number</i>	Number of INVITE retries. Range is from 1 to 10. Default is 6.
---------------------------	---------------	--

Command Default	6 retries
------------------------	-----------

Command Modes	SIP UA 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 relxx	Configures the number of times that the reliable lxx 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 UA configuration commands, with which you configure the user agent.

retry keepalive (SIP)

To set the retry count for keepalive retransmission, use the **retry keepalive** command in SIP UA configuration mode. To restore the retry count to the default value for keepalive retransmission, use the **no** form of this command.

retry keepalive *count*

no retry keepalive *count*

Syntax Description	<i>count</i>	Retry keepalive retransmission value in the range from 1 to 10. The default value is 6.
---------------------------	--------------	---

Command Default The default value for the retry keepalive retransmission is 6.

Command Modes SIP UA configuration

Command History	Release	Modification
	12.4(6)T	This command was introduced.

Usage Guidelines Sets the keepalive retransmissions retry count.

Examples The following example sets the retry for the keepalive retransmissions to 8:

```

sip-ua
  retry keepalive 8

```

Related Commands	Command	Description
	busyout monitor keepalive	Selects a voice port or ports to be busied out in cases of a keepalive failure.
	keepalive target	Identifies a SIP server that will receive keepalive packets from the SIP gateway.
	keepalive trigger	Sets the trigger to the number of Options message requests that must consecutively receive responses from the SIP servers in order to unbusy the voice ports when in the down state.
	timers keepalive	Sets the time interval between sending Options message requests when the SIP server is active or down.

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 UA configuration mode. To reset to the default, use the **no** form of this command.

retry notify *number*

no retry notify

Syntax Description	<i>number</i>	Number of notify message retries. Range is from 1 to 10. Default is 10.
---------------------------	---------------	---

Command Default	10 retries
------------------------	------------

Command Modes	SIP UA 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 UA configuration mode. To reset to the default, use the **no** form of this command.

retry prack *number*

no retry prack

Syntax Description	<i>number</i>	Number of PRACK retries. Range is from 1 to 10. Default is 10.
---------------------------	---------------	--

Command Default	10 retries
------------------------	------------

Command Modes	SIP UA 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 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.
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 UA configuration mode. To reset to the default, use the **no** form of this command.

retry refer *number*

no retry refer

Syntax Description	<i>number</i>	Number of Refer request retries. Range is from 1 to 10. Default is 10.
---------------------------	---------------	--

Command Default	10 retries
------------------------	------------

Command Modes	SIP UA 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 register

To set the total number of Session Initiation Protocol (SIP) Register messages that the gateway should send, use the **retry register** command in SIP UA configuration mode. To reset this number to the default, use the **no** form of this command.

retry register *retries*

no retry register

Syntax Description	<i>retries</i>	Total number of Register messages that the gateway should send. Range is 1 to 10. Default is 10.
---------------------------	----------------	--

Command Default	10 retries
------------------------	------------

Command Modes	SIP UA configuration
----------------------	----------------------

Command History	Release	Modification
	12.2(15)ZJ	This command was introduced.
	12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines	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 specifies that the gateway sends nine Register messages.

```

sip-ua
 retry invite 9
 retry register 9
 timers register 150

```

Related Commands	Command	Description
	registrar	Enables SIP gateways to register E.164 numbers on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and SCCP phones with an external SIP proxy or SIP registrar.
	timers register	Sets how long the SIP UA waits before sending register requests.

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 UA configuration mode. To reset to the default, use the **no** form of this command.

retry rel1xx *number*

no retry rel1xx

Syntax Description	<i>number</i>	Number of reliable 1xx retries. Range is from 1 to 10. Default is 6.
--------------------	---------------	--

Command Default	6 retries
-----------------	-----------

Command Modes	SIP UA 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 1xx 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 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 UA configuration mode. To reset to the default, use the **no** form of this command.

retry response *number*

no retry response

Syntax Description	<i>number</i>	Number of response retries. Range is from 1 to 10. Default is 6.
--------------------	---------------	--

Command Default	6 retries
-----------------	-----------

Command Modes	SIP UA 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 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 subscribe

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

retry subscribe *number*

no retry subscribe *number*

Syntax Description	<i>number</i>	Number of SUBSCRIBE retries. Range is 1 to 10. Default is 10.
---------------------------	---------------	---

Command Default	10 retries
------------------------	------------

Command Modes	SIP UA configuration
----------------------	----------------------

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

Usage Guidelines	Use the retry timer command to configure retry intervals for this command. The default value for retry timer is 1000 ms, and the range is 10 to 100. Setting the timer to lower values can cause the application to get a failure response more quickly.
-------------------------	--

Examples	The following example sets the number of subscribe retries to 5:
-----------------	--

```

sip-ua
  retry subscribe 5

```

Related Commands	Command	Description
	retry notify	Configures the number of times that the Notify message is resent to the user agent that initiated the Invite request.
	retry timer	Configures the retry interval for resending SIP messages.
	show sip-ua retry	Displays SIP user agent retry statistics.

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	<i>window-value</i>	Window value, in minutes. Range is from 1 to 65535. Default is 1440 minutes (24 hours).
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Command Default	1440 minutes (24 hours)
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Command Modes	Annex G neighbor usage configuration
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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 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 *seconds*

no **retry-delay**

Syntax Description	<i>seconds</i>	Interval, in seconds, between attempts to connect with the settlement provider. Range is from 1 to 600.
---------------------------	----------------	---

Command Default	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	<i>number</i>	Maximum number of connection attempts in addition to the first attempt. Default is 1.
---------------------------	---------------	---

Command Default	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	<p><i>cadence-number</i> Number that determines the ringing cadence. Range is from 0 to 2:</p> <ul style="list-style-type: none"> • 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.
---------------------------	---

Command Default	0
------------------------	---

Command Modes	Interface configuration
----------------------	-------------------------

Command History	<table border="1"> <thead> <tr> <th>Release</th> <th>Modification</th> </tr> </thead> <tbody> <tr> <td>12.0(3)T</td> <td>This command was introduced on the Cisco 800 series.</td> </tr> </tbody> </table>	Release	Modification	12.0(3)T	This command was introduced on the Cisco 800 series.
Release	Modification				
12.0(3)T	This command was introduced on the Cisco 800 series.				

Usage Guidelines	<p>This command applies to Cisco 800 series routers.</p> <p>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, see to the <i>Cisco 800 Series Routers Software Configuration Guide</i>.</p>
-------------------------	--

Examples	<p>The following example specifies the type 1 distinctive ring:</p> <pre>ring 1</pre>
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Related Commands	<table border="1"> <thead> <tr> <th>Command</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>destination-pattern</td> <td>Specifies the prefix, the full E.164 telephone number, or an ISDN directory number to be used for a dial peer.</td> </tr> <tr> <td>dial-peer voice</td> <td>Enters dial peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.</td> </tr> <tr> <td>no call-waiting</td> <td>Disables call waiting.</td> </tr> </tbody> </table>	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								
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	
<i>pattern-number</i>	<p>Predefined ring cadence patterns. Each pattern specifies a ring-pulse time and a ring-interval time.</p> <ul style="list-style-type: none"> 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	<p>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.</p>
<i>pulse</i>	<p>Number (1 or 2 digits) specifying ring-pulse (on) time in hundreds of milliseconds.</p> <p>Range is from 1 to 50, for pulses of 100 to 5000 ms. For example: 1 = 100 ms; 10 = 1 s, 40 = 4 s.</p>
<i>interval</i>	<p>Number (1 or 2 digits) specifying ring-interval (off) time in hundreds of milliseconds.</p> <p>Range is from 1 to 50, for pulses of 100 to 5000 ms. For example: 1 = 100 ms; 10 = 1 s, 40 = 4 s.</p>

Command Default 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 2 seconds off on voice port 1/0/0:

```
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 *hertz*

no ring frequency *hertz*

Syntax Description	<i>hertz</i>	Ring frequency, in hertz, used in the FXS interface. Valid entries are as follows: <ul style="list-style-type: none"> Cisco 3600 series: 25 and 50. Default is 25.
---------------------------	--------------	---

Command Default	Cisco 3600 series routers: 25 Hz
------------------------	----------------------------------

Command Modes	Voice-port configuration
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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 voice port to 25 Hz:

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

Related Commands	Command	Description
	ring cadence	Specifies the ring cadence for an FXS voice port.
	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	<i>number</i>	Number of rings detected before answering the call. Range is from 1 to 10. The default is 1.
---------------------------	---------------	--

Command Default	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	<p>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.</p> <p>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.</p> <p>This command is not applicable to Foreign Exchange Station (FXS) or E&M interfaces because they do not receive ringing on incoming calls.</p>
-------------------------	--

Examples	The following example 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
```

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.

Command Default 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 <i>provider-number</i> .
	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.

Command Default 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)	Enables the roaming capability for the dial peer.
	settle-call	Limits the dial peer to using only the specific clearinghouse identified by the specified <i>provider-number</i> .
	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.

Command Default 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.
	12.3(4)T	The default change implemented in Cisco IOS Release 12.3(3) was integrated in Cisco IOS Release 12.3(4)T.

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.

rsvp

To enable RSVP support on a transcoding or MTP device, use the **rsvp** command in DSP farm profile configuration mode. To disable RSVP support, use the **no** form of this command.

rsvp

no rsvp

Syntax Description This command has no arguments or keywords.

Command Default Disabled

Command Modes DSP farm profile configuration

Command History	Release	Modification
	12.4(6)T	This command was introduced.

Usage Guidelines This command enables a transcoder or MTP device to register as RSVP-capable with Cisco Unified CallManager. The SCCP device acts as an RSVP agent under the control of Cisco Unified CallManager. To support RSVP, you must also enable the **codec pass-through** command.



Note This command is not supported in conferencing profiles.

Examples The following example enables RSVP support on the transcoding device defined by profile 200:

```
Router(config)# dspfarm profile 200 transcode
Router(config-dspfarm-profile)# rsvp
Router(config-dspfarm-profile)# codec pass-through
```

Related Commands	Command	Description
	codec (DSP Farm profile)	Specifies the codecs supported by a DSP farm profile.
	debug call rsvp-sync events	Displays events that occur during RSVP setup.
	dspfarm profile	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.
	show sccp connections rsvp	Displays information about active SCCP connections that use RSVP.

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 voice 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-codec-gsmamrnb number | cisco-codec-ilbc
number | cisco-codec-video-h263+ number | cisco-codec-video-h264 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 { cisco-cas-payload number | cisco-clear-channel number |
cisco-codec-fax-ack number | cisco-codec-fax-ind number | cisco-codec-gsmamrnb number |
cisco-codec-ilbc number | cisco-codec-video-h263+ number | cisco-codec-video-h264 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 }]
```

Syntax Description

cisco-cas-payload <i>number</i>	Cisco channel-associated signaling (CAS) RTP payload. Range: 96 to 127. Default: 123.
cisco-clear-channel <i>number</i>	Cisco clear-channel RTP payload. Range: 96 to 127. Default: 125.
cisco-codec-fax-ack <i>number</i>	Cisco codec fax acknowledge. Range: 96 to 127. Default: 97.
cisco-codec-fax-ind <i>number</i>	Cisco codec fax indication. Range: 96 to 127. Default: 96.
cisco-codec-gsmamrnb <i>number</i>	Cisco Global System for Mobile Adaptive Multi-Rate Narrow Band (GSMAMR-NB) codec. Range is from 96 to 127. Default is 117.
cisco-codec-ilbc <i>number</i>	Cisco internet Low Bitrate Codec (iLBC) codec. Range: 96 to 127. Default: 116.
cisco-codec-video-h263+	RTP video codec H.263+ payload type. Range: 96 to 127. Default: 118.
cisco-codec-video-h264	RTP video codec H.264 payload type. Range: 96 to 127. Default: 119.
cisco-fax-relay <i>number</i>	Cisco fax relay. Range: 96 to 127. Default: 122.
cisco-pcm-switch-over-alaw <i>number</i>	Cisco RTP pulse code modulation (PCM) codec switch over indication (a-law). Default: 8.
cisco-pcm-switch-over-ulaw <i>number</i>	Cisco RTP PCM codec switch over indication (mu-law). Default: 0.
cisco-rtp-dtmf-relay <i>number</i>	Cisco RTP dual-tone multifrequency (DTMF) relay. Range: 96 to 127. Default: 121.
nte <i>number</i>	A named telephone event (NTE). Range: 96 to 127. Default: 101.
nse <i>number</i>	A named signaling event (NSE). Range: 96 to 117. Default: 100.

comfort-noise {13 | 19} (Optional) RTP payload type of comfort noise. The July 2001 draft entitled *RTP Payload for Comfort Noise*, from the Internet Engineering Task Force (IETF) Audio/Video Transport (AVT) working group, designates 13 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.

Note This command option is not available on the Cisco AS5400 running NextPort digital signal processors (DSPs). This command option is available on the Cisco AS5400 only if the platform has a high-density packet voice/fax feature card (AS5X-FC) with one or more AS5X-PVDM2-64 DSP modules installed. This support was added in Cisco IOS Release 12.4(4)XC, and integrated into Release 12.4(9)T, and later 12.4T releases.

Command Default No RTP payload type is configured.

Command Modes Dial peer voice configuration (config-dial-peer)

Command History	Release	Modification
	12.2(2)T	This command was introduced.
	12.2(2)XB	The n te and comfort-noise keywords were added.
	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.
	12.4(4)XC	The cisco-codec-gsmamrnb keyword was added.
	12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.
	12.4(11)T	The cisco-codec-ilbc , cisco-codec-video-h263+ , and cisco-codec-video-h264 keywords were added.

Usage Guidelines Use this command to identify the payload type of an RTP. Use this command after the **dtmf-relay** command is used to choose the NTE method of DTMF relay for a Session Initiation Protocol (SIP) call. Configured payload types of NSE and NTE exclude certain values that have been previously hard-coded with Cisco-proprietary meanings. Do not use the following numbers, which have preassigned values: 96, 97, 100, 117, 121 to 123, and 125 to 127.

Use of these values results in an error message when the command is entered. You must first reassign the value in use to a different unassigned number, for example:

```
rtp payload-type cisco-codec-ilbc 100
ERROR: value 100 in use!
```

```
rtp payload-type nse 105
rtp payload-type cisco-codec-ilbc 100
```

Examples

The following example identifies the RTP payload type as GSMAMR-NB124:

```
Router(config-dial-peer)# rtp payload-type cisco-codec-gsmamrnb 124
```

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

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

The following example identifies the RTP payload type for the iLBC as 100:

```
Router(config-dial-peer)# rtp payload-type cisco-codec-ilbc 100
```

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	<i>minutes</i>	Duration, in minutes, to keep the record. Range is from 1 to 10000. Default is 10.
--------------------	----------------	--

Command Default	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

<i>number</i>	Number of records to retain in a session history. Range is from 1 to 100000. Default is 50.
---------------	---

Command Default

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	<i>seconds</i>	How long, in seconds, the router waits to connect to the server before timing out. Range is 1 to 20.
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Command Default	3 seconds
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Command Modes	Global configuration
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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.
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Examples	The following example sets the connection timeout to 10 seconds: <pre>rtsp client timeout connect 10</pre>
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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	<i>seconds</i>	How long, in seconds, the router waits for a response from the server after making a request. Range is 1 to 20.						
Command Default	3 seconds							
Command Modes	Global configuration							
Command History	<table border="1"> <thead> <tr> <th>Release</th> <th>Modification</th> </tr> </thead> <tbody> <tr> <td>12.2(11)T</td> <td>This command was introduced.</td> </tr> </tbody> </table>	Release	Modification	12.2(11)T	This command was introduced.			
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	<p>The following example sets the request timeout to 10 seconds:</p> <pre>rtsp client timeout message 10</pre>							
Related Commands	<table border="1"> <thead> <tr> <th>Command</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>rtsp client session history records</td> <td>Sets the maximum number of records to store in the RTSP client session history.</td> </tr> <tr> <td>rtsp client timeout connect</td> <td>Sets the number of seconds allowed for the router to establish a TCP connection to an RTSP server.</td> </tr> </tbody> </table>	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.	
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 (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 lmatch-pattern lreplacement-rule ldomain-name*

no rule *rule-number preference lmatch-pattern lreplacement-rule ldomain-name*

Syntax	Description
<i>rule-number</i>	Assigns an identification number to the rule. Range is from 1 to 2147483647.
<i>preference</i>	Assigns a preference value to the rule. Range is from 1 to 2147483647. Lower values have higher preference.
<i>lmatch-pattern</i>	Stream editor (SED) expression used to match incoming call information. The slash “/” is a delimiter in the pattern.
<i>lreplacement-rule</i>	SED expression used to replace match-pattern in the call information. The slash “/” is a delimiter in the pattern.
<i>ldomain-name</i>	Domain name to be used while the query to the DNS server is sent.

Command Default No default behavior or values

Command Modes ENUM configuration

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

Usage Guidelines [Table 31](#) shows examples of match patterns, input strings, and result strings for the rule (voice translation-rule) command.

Table 31 Match Patterns, Input Strings and Result Strings

Match Pattern	Replacement Pattern	Input String	Result String	Description
/^.* /	//	4085550100	–	Any string to null string.
/^456\ (.*) /	/555\1 /	5550100	5550100	Match from the beginning of the input string.
/\ (^...\)456\ (...)\ /	/\1555\2 /	408555010	4085550100	Match from the middle of the input string.
/\ (.*)0100 /	/\0199 /	4085550100	4085550199	Match from the end of the input string.

Table 31 Match Patterns, Input Strings and Result Strings (continued)

Match Pattern	Replacement Pattern	Input String	Result String	Description
/^1#\(.*\)/	/\1/	1#2345	2345	Replace match string with null string.
/^408...\(8333\)/	/555\1/	4085550100	5550100	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

<i>precedence</i>	Priority of the translation rule. Range is from 1 to 15.
<i>/match-pattern/</i>	Stream editor (SED) expression used to match incoming call information. The slash '/' is a delimiter in the pattern.
<i>/replace-pattern/</i>	SED expression used to replace the match pattern in the call information. The slash '/' is a delimiter in the pattern.
type <i>match-type replace-type</i>	<p>(Optional) Number type of the call. Valid values for the <i>match-type</i> argument are as follows:</p> <ul style="list-style-type: none"> • 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. <p>Valid values for the <i>replace-type</i> argument are as follows:</p> <ul style="list-style-type: none"> • 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 <i>match-type replace-type</i> (continued)	<ul style="list-style-type: none"> • 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 <i>match-type replace-type</i>	<p>(Optional) Numbering plan of the call. Valid values for the <i>match-type</i> argument are as follows:</p> <ul style="list-style-type: none"> • 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. <p>Valid values for the <i>replace-type</i> argument are as follows:</p> <ul style="list-style-type: none"> • 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.

Command Default 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

Use this command in conjunction after the **voice translation-rule** command. An earlier version of this command uses the same name but is used after 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, we recommend 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.

Table 32 shows examples of match patterns, input strings, and result strings for the rule (voice translation-rule) command.

Table 32 Match Patterns, Input Strings and Result Strings

Match Pattern	Replacement Pattern	Input String	Result String	Description
/^.* /	//	4085550100	—	Any string to null string.
//	//	4085550100	4085550100	Match any string but no replacement. Use this to manipulate the call plan or call type.
/\(^...\)\456\(...)\ /	/\1555\2 /	4084560177	4085550177	Match from the middle of the input string.
/\(...\)\0120 /	/\10155 /	4081110120	4081110155	Match from the end of the input string.
/^1#\(...\)\ /	/\1 /	1#2345	2345	Replace match string with null string.
/^408...\(8333\)\ /	/555\1 /	4087770100	5550100	Match multiple patterns.
/1234 /	/00&00 /	5550100	55500010000	Match the substring.
/1234 /	/00\000 /	5550100	55500010000	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 5550105 or 70105, translation rule 21 uses the rule command to forward the number to 14085550105 instead.

```
Router(config)# voice translation-rule 21
Router(cfg-translation-rule)# rule 1 /^5550105/ /14085550105/
Router(cfg-translation-rule)# rule 2 /^70105/ /14085550105/
```

In the next example, if a called number is either 14085550105 or 014085550105, after the execution of translation rule 345, the forwarding digits are 50105. 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 /^14085550105/ /50105/ plan any national
Router(cfg-translation-rule)# rule 2 /^014085550105/ /50105/ 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.