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## 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

<b>mandatory</b>	Makes the presence of the Service-Type attribute mandatory in RADIUS Access-Accept messages.
<b>on-for-login-auth</b>	Sends the Service-Type attribute in the authentication packets.  <b>Note</b> 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.
<b>support-multiple</b>	Supports multiple Service-Type values for each RADIUS profile.
<b>voice</b> <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 <b>mandatory</b> keyword was added.
12.2SX	This command is supported in the Cisco IOS Release 12.2SX train. Support in a specific 12.2SX release of this train depends on your feature set, platform, and platform hardware.

### 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
```

# rai target

To configure the Session Initiation Protocol (SIP) Resource Allocation Indication (RAI) mechanism, use the **rai target** command in SIP UA configuration mode. To disable SIP RAI configuration, use the **no** form of this command.

```
rai target target-address resource-group group-index [transport [{tcp [tls [scheme {sip | sips}] | udp}] ]
```

```
no rai target target-address
```

## Syntax Description

<i>target-address</i>	IPv4, IPv6, or Domain Name Server (DNS) target address to which the status of the gateway resources are reported. The format of the target address can be one of the following: <ul style="list-style-type: none"> <li>• <b>ipv4:</b> <i>ipv4-address</i></li> <li>• <b>ipv6:</b> <i>ipv6-address</i></li> <li>• <b>dns:</b> <i>domain-name</i></li> </ul>
<b>resource-group</b>	Maps the target address with the resource group index.
<i>group-index</i>	Resource group index. The range is from 1 to 5.
<b>transport</b>	(Optional) Specifies the mechanism to transport the RAI information.
<b>tcp</b>	(Optional) Transports the RAI information through Transmission Control Protocol (TCP).
<b>tls</b>	(Optional) Transports the RAI information through Transport Layer Security (TLS).
<b>scheme</b>	(Optional) Specifies the URL scheme for outgoing messages.
<b>sip</b>	(Optional) Selects SIP URL in outgoing OPTIONS message.
<b>sips</b>	(Optional) Selects Secure SIP (SIPS) URL in outgoing OPTIONS message.
<b>udp</b>	(Optional) Transports the RAI information through Unified Datagram Protocol (UDP).

## Command Default

The SIP RAI mechanism is disabled.

## Command Modes

SIP UA configuration (config-sip-ua)

## Command History

Release	Modification
15.1(2)T	This command was introduced.

## Usage Guidelines

Use the **rai target** command to provide the details of SIP along with the index of the resource group that needs to be monitored for reporting over SIP trunk. A maximum of five RAI configurations can be applied for other destination targets or monitoring entities. However, only one RAI configuration is possible for one target address.

## Examples

The following example shows how to enable reporting of SIP RAI information over TCP to a target address of example.com:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# rai target dns:example.com resource-group 1
```

## Related Commands

Command	Description
<b>debug rai</b>	Enables debugging for Resource Allocation Indication (RAI).
<b>periodic-report interval</b>	Configures periodic reporting parameters for gateway resource entities.
<b>resource (voice)</b>	Configures parameters for monitoring resources, use the resource command in voice-class configuration mode.
<b>show voice class resource-group</b>	Displays the resource group configuration information for a specific resource group or all resource groups.
<b>voice class resource-group</b>	Enters voice-class configuration mode and assigns an identification tag number for a resource group.

# random-contact

To populate an outgoing INVITE message with random-contact information (instead of clear-contact information), use the **random-contact** command in voice service VoIP SIP configuration mode or voice class tenant configuration mode. To disable random-contact information, use the **no** form of this command.

**random-contact system**  
**no random-contact**

## Syntax Description

<b>system</b>	Specifies that the random-contact information use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.
---------------	---

## Command Default

Outgoing INVITE messages are populated with clear-contact information.

## Command Modes

Voice service VoIP SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

## Command History

Release	Modification
12.4(22)YB	This command was introduced.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models.

## Usage Guidelines

To populate outbound INVITE messages from the Cisco Unified Border Element with random-contact information instead of clear-contact information, use the **random-contact** command. This functionality will work only when the Cisco Unified Border Element is configured for Session Initiation Protocol (SIP) registration with random contact using the **credentials** and **registrars** commands.

## Examples

The following example shows how to populate outbound INVITE messages with random-contact information:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# random-contact
```

The following example shows how to populate outbound INVITE messages with random-contact information:

```
Router(config-class)# random-contact system
```



**Related Commands**

<b>Command</b>	<b>Description</b>
credentials (sip ua)	Sends a SIP registration message from a Cisco Unified Border Element in the UP state.
registrar	Enables SIP gateways to register E.164 numbers on behalf of FXS, EFXS, and SCCP phones with an external SIP proxy or SIP registrar.
voice-class sip random-contact	Populates the outgoing INVITE message with random-contact information at the dial-peer level.

## random-request-uri validate

To enable the validation of the called number based on the random value generated during the registration of the number, use the **random-request-uri validate** command in voice service VoIP SIP configuration mode or voice class tenant configuration mode. To disable validation, use the **no** form of this command.

**random-request-uri validate system**  
**no random-request-uri validate**

<b>Syntax Description</b>	<b>system</b>	Specifies that the validated called number use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.
---------------------------	---------------	--

**Command Default** Validation is disabled.

**Command Modes** Voice service voip sip configuration (conf-serv-sip)  
 Voice class tenant configuration (config-class)

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.4(22)YB	This command was introduced.
	15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.

**Usage Guidelines** The system generates a random string when registering a new number. An INVITE message with the P-Called-Party-ID value can have the Request-URI set to this random number. To enable the system to identify the called-number from the random number in the Request-URI, use the **random-request-uri validate** command.

If the P-Called-Party-ID is not set in the INVITE message, the Request URI for that message must contain the called party information (and cannot contain a random number). Therefore validation is performed only on INVITE messages with a P-Called-Party-ID.

### Examples

The following example shows how to enable called-number validation at the global configuration level:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# random-request-uri validate
```

The following example shows how to enable called-number validation in the voice class tenant configuration mode:

```
Router(config-class)# random-request-uri validate system
```

Related Commands	Command	Description
	credentials (sip ua)	Sends a SIP registration message from a Cisco Unified Border Element in the UP state.
	register	Enables SIP gateways to register E.164 numbers on behalf of FXS, EFXS, and SCCP phones with an external SIP proxy or SIP registrar.
	<b>voice-class sip random-request-uri validate</b>	Validates the called number based on the random value generated during the registration of the number at the dial-peer configuration level.

## 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

<b>all</b>	Configures all RAS message counters that do not have explicit values configured individually. If <b>no ras retry all</b> is entered, all values are set to the default except for the individual values that were configured separately.
<b>arq</b>	Configures the admission request (ARQ) message counter.
<b>brq</b>	Configures the bandwidth request (BRQ) message counter.
<b>drq</b>	Configures the disengage request (DRQ) message counter.
<b>grq</b>	Configures the gatekeeper request (GRQ) message counter.
<b>rai</b>	Configures the resource availability indication (RAI) message counter.
<b>rrq</b>	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 <b>ras timeout</b> 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**

<b>Command</b>	<b>Description</b>
<b>ras timeout</b>	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

<b>lrq</b>	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 <b>ras timeout lrq</b> 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
<b>ras timeout lrq</b>	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

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
```

Command	Description
<b>rrq dynamic -prefixes-accept</b>	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

<code>time-to-live seconds</code>	Number of seconds that the gatekeeper should consider the gateway active. Valid values are 15 through 4000. The time-to-live seconds value must be greater than the margin seconds value.
<b>margin seconds</b>	(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 time value times two must be less than or equal to the time-to-live seconds 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 time-to-live value was changed from 300 to 4000 seconds.
12.3(4)T2	The maximum time-to-live value was changed from 300 to 4000 seconds.
12.3(7)T	The maximum time-to-live value was changed from 300 to 4000 seconds.

### Usage Guidelines

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

### Examples

The following example shows the `time-to-live 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

<b>all</b>	Configures message timeout values for all RAS messages that do not have explicit values configured individually. If no <code>ras timeout all</code> is entered, all values are set to the default except for the individual values that were configured separately.
<b>arq</b>	Configures the admission request (ARQ) message timer.
<b>brq</b>	Configures the bandwidth request (BRQ) message timer.
<b>drq</b>	Configures the disengage request (DRQ) message timer.
<b>grq</b>	Configures the gatekeeper request (GRQ) message timer.
<b>rai</b>	Configures the resource availability indication (RAI) message timer.
<b>rrq</b>	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
<b>ras retry</b>	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		
<b>all</b>		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.
<b>arq</b>		Configures the admission request (ARQ) message timer. Default: 3.
<b>brq</b>		Configures the bandwidth request (BRQ) message timer. Default: 3.
<b>drq</b>		Configures the disengage request (DRQ) message timer. Default: 3.
<b>grq</b>		Configures the gatekeeper request (GRQ) message timer. Default: 5.
<b>rai</b>		Configures the resource availability indication (RAI) message timer. Default: 3.
<b>rrq</b>		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
<b>ras retry</b>	Configures the H.323 RAS message retry counters.
<b>ras timeout</b>	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

<b>lrq</b>	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
<b>ras retry lrq</b>	Configures the gatekeeper RAS message retry counters.

## rbs-zero

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

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

### Syntax Description

<b>nfas-int</b>	<i>nfas-int-range</i>	(Optional) Non-Facility Associated Signaling (NFAS) interface number. Range is from 0 to 32.
-----------------	-----------------------	--

### Command Default

IAESS 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 IAESS switches for dial-in and dial-out calls. Modem calls of 56K or a lower rate are accepted; 64K calls are rejected.

In IAESS mode, the following occurs:

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

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

### Examples

The following example enables IAESS 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**

<b>Command</b>	<b>Description</b>
<b>interface serial</b>	Enters serial interface configuration mode.
<b>isdn switch -type</b>	Sets the switch type.
<b>pri -group timeslots</b>	Configures the PRI trunk for a designated operation.
<b>show controllers t1</b>	Displays information about the T1 links and the hardware and software driver information for the T1 controller.
<b>show isdn nfas group</b>	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 or voice class tenant configuration mode. To disable reason-header override, use the **no** form of this command.

**reason-header override system**  
**no reason-header override system**

<b>Syntax Description</b>	<b>system</b>	Specifies that the override header use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.
---------------------------	---------------	--

**Command Default** No default behavior or values.

**Command Modes** SIP UA configuration  
 Voice class tenant configuration (config-class)

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG entry.

**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
```

The following example, shows the SIP user agent with reason-header override being configured in the voice class tenant configuration mode:

```
Router(config-class)# reason-header override system
```

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>sip-ua</b>	Enables SIP UA configuration commands.



## record-entry

To specify the trustpoints to be used for the creation of the Cisco Certificate Trust List (CTL) file, use the **record-entry** command in CTL file configuration mode. To remove a record entry from a CTL, use the **no** form of the command.

```
record-entry {capf | cucm-tftp | selfsigned} trustpoint trustpoint-name
no record-entry {capf | cucm-tftp | selfsigned} trustpoint trustpoint-name
```

Syntax Description		
<b>capf</b>		Specifies that the trustpoint is created using the CAPF certificate imported from Cisco Unified Communications Manager to the device.
<b>cucm-tftp</b>		Specifies the role of this trustpoint to be Cisco Unified Call Manager and TFTP.
<b>selfsigned</b>		Specifies that the trustpoint is self-signed by the router.
<b>trustpoint</b> <i>trustpoint-name</i>		Specifies the name of the trustpoint.

**Command Default** No trustpoints are specified for the CTL file.

**Command Modes** CTL file configuration mode (config-ctl-file)

**Command History**

Release	Modification
15.3(3)M	This command was introduced.

### Usage Guidelines

#### Example

The following example shows how to specify that the trustpoint is created using the CAPF certificate imported from CUCM. The trustpoint is called “trustpoint\_1”:

```
Device(config)# voice-ctl-file myctl
Device(config-ctl-file)# record-entry capf trustpoint trustpoint_1
```

# recorder profile

To configure a media profile recorder, use the **recorder profile** command in media class configuration mode. To disable the configuration, use the **no** form of this command.

**recorder profile** *tag*  
**no recorder**

## Syntax Description

<i>tag</i>	Media profile recorder tag. The range is from 1 to 10000.
------------	---

## Command Default

A media profile recorder is not configured.

## Command Modes

Media class configuration (cfg-mediaclass)

## Command History

Release	Modification
15.1(2)T	This command was introduced.

## Usage Guidelines

Use the **recorder profile** command to associate a recorder profile with a media class. The configured recorder profile specifies the recorder profile that is used by the media class. You can configure any number of recorder profiles.

## Examples

The following example shows how to configure a media profile recorder:

```
Router# configure terminal
Router(config) media class 200
Router(cfg-mediaclass)# recorder profile 100
```

## Related Commands

Command	Description
<b>media class</b>	Enters media class configuration mode.

# redial

To define speed-dial code for a Feature Speed-dial (FSD) 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 redial** form of this command.

**redial** *keypad-character*  
**no redial**

Syntax Description	<i>keypad-character</i>
	Character string that can be dialed on a telephone keypad (0-9, *, #). Default: #. Before Cisco IOS Release 12.4(20)YA, this is a single character. In Cisco IOS Release 12.5(20)YA and later releases, the string can be any of the following: <ul style="list-style-type: none"> <li>• A single character (0-9, *, #)</li> <li>• Two digits (00-99)</li> <li>• Two to four characters (0-9, *, #) and the leading or ending character must be an asterisk (*) or number sign (#)</li> </ul>

**Command Default** The default value is # (number sign).

**Command Modes** STC application feature speed-dial configuration (config-stcapp-fsd)

Command History	Release	Modification
	12.4(2)T	This command was introduced.
	12.4(20)YA	The length of the <i>keypad-character</i> argument was changed to 1 to 4 characters.
	12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.

**Usage Guidelines** This command changes the value of the speed-dial code for Redial from the default (#) to the specified value.

In Cisco IOS Release 12.4(20)YA and later releases, if the length of the *keypad-character* argument is at least two characters and the leading or ending character of the string is an asterisk (\*) or a number sign (#), phone users are not required to dial a prefix to access this speed dial. Typically, phone users dial a Feature Speed-dial (FSD) consisting of a prefix plus a speed-dial code, for example \*#. If the feature code is 78#, the phone user dials only 78#, without the FSD prefix, to access the corresponding feature.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that is already being used for a feature access code (FAC) or another FSD, you receive a 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.

In Cisco IOS Release 12.4(20)YA and later releases, if you attempt to configure this command with a value that precludes or is precluded by a feature code for a FAC or another FSD, you receive a message. If you configure this command with a value that precludes or is precluded by another code, the system always executes the call feature with the shortest code and ignores the longer code. For example, #1 will always

preclude #12 and #123. You must configure a new value for the precluded code in order to enable access to that feature.

To display a list of all FACs and FSDs, use the **show stcapp feature codes** command.

### Examples

The following example shows how to change the value of the speed-dial code for Redial from the default (#). In this configuration, a phone user must press \*\* on the keypad to redial the number that was most recently dialed on this line, regardless of what value is configured for the FSD prefix.

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

### Related Commands

Command	Description
<b>digit</b>	Designates the number of digits for feature speed-dial codes (FSDs).
<b>prefix (stcapp-fsd)</b>	Defines the prefix for feature speed-dials (FSDs).
<b>show stcapp feature codes</b>	Displays all feature access codes (FACs) and feature access codes (FSDs) that are available for the STC application.
<b>speed dial</b>	Designates a range of speed-dial codes for the STC application.
<b>stcapp feature speed-dial</b>	Enables feature speed-dials (FSDs) in STC application and enters STC application feature speed-dial configuration mode for changing values of the prefix and speed-dial codes from the default.

## 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
	<b>sip</b>	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
<b>redirect ip2ip (voice service)</b>	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.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

**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
	<b>redirect ip2ip (dial peer)</b>	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 or voice class tenant configuration mode. To disable the handling of 3xx redirect messages, use the **no** form of this command.

**redirection system**  
**no redirection system**

### Syntax Description

<b>system</b>	Specifies that the SIP redirection messages use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.
---------------	---

### Command Default

Redirection is enabled.

### Command Modes

SIP UA configuration  
 Voice class tenant configuration (config-class).

### Command History

Release	Modification
12.2(13)T	This command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

### 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
```

The following example enables processing of incoming 3xx redirection messages in the voice class tenant configuration mode:

```
Router(config-class)# redirection system
```



**Related Commands**

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

# redundancy-reload

To reload control when the redundancy group (RG) fails, use the **redundancy-reload** command in global VoIP configuration mode. To enable the device to transition into PROTECTED mode (high availability), use the **no** form of this command.

**redundancy-reload**  
**no redundancy-reload**

**Syntax Description** This command has no arguments or keywords.

**Command Default** The PROTECTED mode for voice high availability is not enabled.

**Command Modes** Global VoIP configuration (conf-voi-serv)

Command History	Release	Modification
	Cisco IOS XE Release 3.11S	This command was introduced.

**Usage Guidelines** Use the **no redundancy-reload** command to enable the device to transition into PROTECTED mode. The default form of this command is **redundancy-reload**.

In the PROTECTED mode:

- Bulk synchronization request, call checkpointing, and incoming call processing are disabled.
- The device needs to be manually reloaded to exit from this state.

**Examples** The following example enables the PROTECTED mode for the device:

```
Device(config)# voice service voip
Device(conf-voi-serv)#no redundancy-reload
```

# redundancy group

To associate the interface with the redundancy group created, use the **redundancy group** command in interface mode. To dissociate the interface, use the **no** form of this command.

**redundancy group** *group-number* { **ipv4** | **ipv6** } *ip address* **exclusive**  
**no redundancy group** *group-number* { **ipv4** | **ipv6** } *ip address* **exclusive**

Syntax Description		
	<i>group-number</i>	Specifies the redundancy group number.
	<i>ip address</i>	Specifies IPv4 or IPv6 address.
	<b>exclusive</b>	Associates the redundancy group to the interface.

**Command Default** No default behavior or values

**Command Modes** Interface configuration mode (config-if)

Command History	Release	Modification
	Cisco IOS XE Dublin 17.12.1a	This command was introduced.

**Usage Guidelines** You can configure a maximum of two redundancy groups. Hence, there can be only two Active and Standby pairs within the same network.

## Examples

The following example configuration shows how to associate the IPv4 interface with the redundancy group:

```
Router(config-if)# redundancy group 1 ip 10.64.86.126 exclusive
```

The following example configuration shows how to associate the IPv6 interface with the redundancy group:

```
Router(config-if)# redundancy group 1 ipv6 2001:10:64:86::126/119 exclusive
```

Related Commands	Command	Description
	<b>ipv6 address</b> <i>ip-address</i>	Physical IPv6 address configuration of the device.

# refer-delay-disconnect

To delay the disconnect on transferor leg after successful transfer completion, use the **refer-delay-disconnect** command. If the call leg is not disconnected within the specified timeout, CUBE disconnects the call leg with BYE message.

**refer-delay-disconnect** <1-5>  
**no refer-delay-disconnect**

<1-5>	Specifies that CUBE delays the disconnect message (sending BYE) on the transferor leg for the configured timeout.
-------	---

**Command Default** Refer-delay-disconnect is disabled.

**Command Modes** Voice service voip SIP configuration (conf-serv-sip)  
 Voice class tenant configuration (config-class)  
 Dial peer configuration

Command History	Release	Modification
	Cisco IOS XE Bengaluru 17.6.1a	This command is introduced.

**Usage Guidelines** When this command is configured, CUBE delays the disconnect message on transferor leg for the configured time. Default value is not enabled. Hence without this config, CUBE disconnects the call immediately after REFER transaction completion.

**Examples** The following example shows how to enable refer-delay-disconnect on the CUBE in voice service configuration mode:

```
Router(config)# voice service voip
  Router(conf-voi-serv)#sip
  Router(conf-serv-sip)#refer-delay-disconnect 3
```

**Examples** The following example shows how to enable refer-delay-disconnect on the CUBE in the voice class tenant configuration mode:

```
Router(config)# voice class tenant 10
  Router(config-class)#refer-delay-disconnect 3
```

**Examples** The following example shows how to enable refer-delay-disconnect on the CUBE in dial-peer configuration mode:

```
Router(config)#dial-peer voice 22 voip
  Router(config-dial-peer)#voice-class sip refer-delay-disconnect 3
```

**Related Commands**

Command	Description
<b>refer-delay-disconnect (dial peer)</b>	Delays the disconnect message on a transferor leg with a BYE message on a specific VoIP dial peer using CUBE .

## refer-ood enable

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

**refer-ood enable** [*request-limit*] [**system**]  
**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.
<b>system</b>	Specifies that the out-of-dialog refer (OOD-R) processing use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

### Command Default

OOD-R processing is disabled.

### Command Modes

SIP UA configuration (config-sip-ua)

Voice class tenant configuration (config-class)

### Command History

Release	Cisco product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
15.6(2)T and IOS XE Denali 16.3.1	CUBE	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.

### 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.

### Examples

The following example shows how to enable OOD-R:

```
Router(config)# sip-ua
Router(config-sip-ua)# refer-ood enable
```

The following example shows how to enable OOD-R in the voice class tenant configuration mode:

```
Router(config-class)# refer-ood enable system
```

**Related Commands**

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

## referto-passing

To disable dial peer lookup and modification of the Refer-To header when the Cisco Unified Border Element (UBE) passes across a REFER message during a call transfer, use the **referto-passing** command in voice service voip SIP configuration mode or voice class tenant configuration mode. To enable dial peer lookup and the Refer-To header modification, use the **no** form of this command.

**referto-passing system**

**no referto-passing system**

### Syntax Description

<b>system</b>	Specifies that the enable dial peer lookup and the Refer-To header modification use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.
---------------	---

### Command Default

Dial peer lookup is performed. The Refer-To header is modified to include the address of the CUBE if address hiding is enabled or to include the address of the call target if a dial peer match is found.

### Command Modes

Voice service voip SIP configuration (conf-serv-sip).

Voice class tenant configuration (config-class).

### Command History

Release	Modification
15.2(1)T	This command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

### Usage Guidelines

By default, while passing across the REFER message, the CUBE replaces the host portion of the Refer-To header with the address of the CUBE if the **address-hiding** command is enabled or with the address of the call target if a dial peer match is found. You can use the **referto-passing** command to disable the CUBE from overwriting the Refer-To header even if address hiding is enabled. This command also disables dial peer lookup when the CUBE passes across the REFER message.

### Examples

The following example shows how to enable REFER message pass-through on the CUBE and disable the modification of the Refer-To header:

```
Router(config)# voice service voip
Router(conf-voi-serv)# supplementary-service sip refer
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# referto-passing
```

The following example shows how to enable REFER message pass-through on the CUBE in the voice class tenant configuration mode:

```
Router(config-class)# referto-passing system
```



**Related Commands**

<b>Command</b>	<b>Description</b>
<b>address-hiding</b>	Hides signaling and media peer addresses from endpoints other than the gateway.
<b>sip</b>	Enters SIP configuration mode from voice service voip configuration mode.
<b>supplementary-service sip refer</b>	Enables REFER message pass-through on the CUBE.

## 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 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
<b>destination -pattern</b>	Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer.
<b>dial -peer (voice)</b>	Enters dial peer configuration mode and specifies the method of voice encapsulation.
<b>port (dial peer)</b>	Associates a dial peer with a specific voice port.
<b>show gateway</b>	Displays the current gateway status.
<b>zone prefix</b>	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
<b>destination-pattern not-provided</b>	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 Skinny Client Control Protocol (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 {dhcp | [registrar-index] registrar-server-address [: port]} [auth-realm realm] [expires
seconds] [random-contact] [refresh-ratio ratio-percentage] [scheme {sip | sips}] [tcp] [type]
[secondary]
server | {expires | system}
no registrar [{registrar-index | secondary}]
```

Syntax Description	
<b>dhcp</b>	(Optional) Specifies that the domain name of the primary registrar server is retrieved from a DHCP server (cannot be used to configure secondary or multiple registrars).
<i>registrar-index</i>	(Optional) A specific registrar to be configured, allowing configuration of multiple registrars (maximum of six). Range is 1–6.
<i>registrar-server-address</i>	The SIP registrar server address to be used for endpoint registration. This value can be entered in one of three formats: <ul style="list-style-type: none"> <li>• <b>dns:</b> <i>address</i> --the Domain Name System (DNS) address of the primary SIP registrar server (the <b>dns:</b> delimiter must be included as the first four characters).</li> <li>• <b>ipv4:</b> <i>address</i> --the IP address of the SIP registrar server (the <b>ipv4:</b> delimiter must be included as the first five characters).</li> <li>• <b>ipv6:</b>[ <i>address</i> ]--the IPv6 address of the SIP registrar server (the <b>ipv6:</b> delimiter must be included as the first five characters and the address itself must include opening and closing square brackets).</li> </ul>
: <i>port</i> ]	(Optional) The SIP port number (the colon delimiter is required).
<b>auth-realm</b>	(Optional) Specifies the realm for preloaded authorization.
<i>realm</i>	The realm name.
<b>expires</b> <i>seconds</i>	(Optional) Specifies the default registration time, in seconds. Range is 60–65535. Default is 3600.
<b>random-contact</b>	(Optional) Specifies the Random String Contact header that is used to identify the registration session.
<b>refresh-ratio</b> <i>ratio-percentage</i>	(Optional) Specifies the registration refresh ratio, in percentage. Range is 1–100. Default is 80.
<b>scheme</b> { <b>sip</b>   <b>sips</b> }	(Optional) Specifies the URL scheme. The options are SIP ( <b>sip</b> ) or secure SIP ( <b>sips</b> ), depending on your software installation. The default is <b>sip</b> .

<b>tcp</b>	(Optional) Specifies TCP. If not specified, the default is UDP UDP.
<i>type</i>	(Optional) The registration type. <b>Note</b> The <i>type</i> argument cannot be used with the <b>dhcp</b> option.
<b>secondary</b>	(Optional) Specifies a secondary SIP registrar for redundancy if the primary registrar fails. This option is not valid if DHCP is specified.  When there are two registrars, REGISTER message is sent to both the registrar servers, even if the primary registrar sends a 200 OK and the trunk is registered to the primary registrar.  If you want to send the registration to the secondary registrar, only when the primary fails, then use DNS SRV.  <b>Note</b> You cannot configure any other optional settings once you enter the <b>secondary</b> keyword--specify all other settings first.
<b>expires</b>	(Optional) Specifies the registration expiration time
<b>system</b>	(Optional) Specifies the usage of global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default**

Registration is disabled.

**Command Modes**

SIP UA configuration (config-sip-ua)

Voice class tenant configuration (config-class)

**Command History**

<b>Release</b>	<b>Modification</b>
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	This command was modified. The <b>tls</b> keyword and the <b>scheme</b> keyword with the <i>string</i> argument were added.
12.4(22)T	This command was modified. Support for IPv6 addresses was added.
12.4(22)YB	This command was modified. The <b>dhcp</b> , <b>random-contact</b> and <b>refresh-ratio</b> keywords were added. Also, the <b>aor-domain</b> keyword and the <b>tls</b> option for the tcp keyword were removed.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.0(1)XA	This command was modified. The <i>registrar-index</i> argument for support of multiple registrars on SIP trunks was added.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(2)T	This command was modified. The <b>auth-realm</b> keyword was added.

Release	Modification
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

### Usage Guidelines

Use the **registrar dhcp** or **registrar registrar-server-address** command to enable the gateway to register E.164 phone numbers with primary and secondary external SIP registrars. In Cisco IOS Release 15.0(1)XA and later releases, endpoints on Cisco IOS SIP time-division multiplexing (TDM) gateways, Cisco Unified Border Elements (CUBEs), and Cisco Unified Communications Manager Express (Cisco Unified CME) can be registered to multiple registrars using the **registrar registrar-index** command.

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



**Note** When entering an IPv6 address, you must include square brackets around the address value.

### Examples

The following example shows how to configure registration with a primary and secondary registrar:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# retry invite 3
Router(config-sip-ua)# retry register 3
Router(config-sip-ua)# timers register 150
Router(config-sip-ua)# registrar ipv4:209.165.201.1 expires 14400 secondary
```

The following example shows how to configure a device to register with the SIP server address received from the DHCP server. The **dhcp** keyword is available only for configuration by the primary registrar and cannot be used if configuring multiple registrars.

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# registrar dhcp expires 14400
```

The following example shows how to configure a primary registrar using an IP address with TCP:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# retry invite 3
Router(config-sip-ua)# retry register 3
Router(config-sip-ua)# timers register 150
Router(config-sip-ua)# registrar ipv4:209.165.201.3 tcp
```

The following example shows how to configure a URL scheme with SIP security:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# retry invite 3
Router(config-sip-ua)# retry register 3
```

```
Router(config-sip-ua) # timers register 150
Router(config-sip-ua) # registrar ipv4:209.165.201.7 scheme sips
```

The following example shows how to configure a secondary registrar using an IPv6 address:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua) # registrar ipv6:[3FFE:501:FFFF:5:20F:F7FF:FE0B:2972] expires 14400
secondary
```

The following example shows how to configure all POTS endpoints to two registrars using DNS addresses:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua) # registrar 1 dns:example1.com expires 180
Router(config-sip-ua) # registrar 2 dns:example2.com expires 360
```

The following example shows how to configure the realm for preloaded authorization using the registrar server address:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua) # registrar 2 192.168.140.3:8080 auth-realm example.com expires 180
```

The following example shows how to configure registrar in the voice class tenant configuration mode:

```
Router(config-class)# registrar server system
```

## Related Commands

Command	Description
<b>authentication (dial peer)</b>	Enables SIP digest authentication on an individual dial peer.
<b>authentication (SIP UA)</b>	Enables SIP digest authentication.
<b>credentials (SIP UA)</b>	Configures a Cisco UBE to send a SIP registration message when in the UP state.
<b>localhost</b>	Configures global settings for substituting a DNS local host name in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages.
<b>retry register</b>	Sets the total number of SIP register messages to send.
<b>show sip-ua register status</b>	Displays the status of E.164 numbers that a SIP gateway has registered with an external primary or secondary SIP registrar.
<b>timers register</b>	Sets how long the SIP UA waits before sending register requests.
<b>voice-class sip localhost</b>	Configures settings for substituting a DNS localhost name in place of the physical IP address in the From, Call-ID, and Remote-Party-ID headers of outgoing messages on an individual dial peer, overriding the global setting.



# registrar server

To enable the local Session Initiation Protocol (SIP) registrar, use the **registrar server** command in service SIP configuration mode. To disable the configuration, use the **no** form of this command.

```
registrar server [expires [max value] [min value]]
no registrar server
```

Syntax Description	expires	(Optional) Configures the registration expiry time.
	max value	(Optional) Configures the maximum registration expiry time, in seconds. The range is from 120 to 86400. The default is 3600.
	min value	(Optional) Configures the minimum registration expiry time, in seconds. The range is from 60 to 3600. The default is 60.

**Command Default** The local SIP registrar is disabled.

**Command Modes** Service SIP configuration (conf-serv-sip)

Command History	Release	Modification
	15.1(3)T	This command was introduced.
	Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

**Usage Guidelines** You must enable the local SIP registrar by using the **registrar server** command before configuring the SIP registration on Cisco Unified Border Element (UBE).

**Examples** The following example shows how to enable the local SIP registrar and set the maximum and minimum expiry values to 4000 and 100 seconds respectively:

```
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# registrar server expires max 4000 min 100
```

Related Commands	Command	Description
	registration passthrough	Configures SIP registration pass-through options at the global level.
	voice-class sip registration passthrough	Configures SIP registration pass-through options on a dial peer.

## 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**

<b>Syntax Description</b>	<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

<b>Command History</b>	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
```

<b>Related Commands</b>	Command	Description
	<b>ccm group</b>	Creates a Cisco Unified CallManger group and enters SCCP Cisco CallManager configuration mode.
	<b>registration timeout</b>	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**

<b>Syntax Description</b>	<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

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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-ccm) #
  registration timeout 12
```

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

# registration passthrough

To configure the Session Initiation Protocol (SIP) registration pass-through options, use the **registration passthrough** command in service SIP configuration mode or voice class tenant configuration mode. To disable the configuration, use the **no** form of this command.

**registration passthrough** [*static*] [*rate-limit* [*expires value*] [*fail-count value*]] [*registrar-index* [*index*]][*system*]  
**no registration passthrough**

## Syntax Description

<b>static</b>	(Optional) Configures Cisco Unified Border Element (UBE) to use static registrar details for SIP registration. Cisco UBE works in point-to-point mode when the <b>static</b> keyword is used.
<b>rate-limit</b>	(Optional) Configures SIP registration pass-through rate limit options.
<b>expires</b> <i>value</i>	(Optional) Sets the expiry value for rate limiting, in seconds. The range is from 60 to 65535. The default value is 3600.
<b>fail-count</b> <i>value</i>	(Optional) Sets the fail count value for rate limiting. The range is from 2 to 20. The default value is 0.
<b>registrar-index</b>	(Optional) Configures the registrar index that is to be used for registration pass-through.
<i>index</i>	(Optional) Registration index value. The range is from 1 to 6.
<b>system</b>	Specifies that the registration pass-through options use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

## Command Default

SIP registration pass-through options are not configured.

## Command Modes

Service SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

## Command History

Release	Modification
15.1(3)T	This command was introduced.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.

## Usage Guidelines

You can use the **registration passthrough** command to configure the following SIP pass-through functionalities:

- Back-to-back registration facility to register phones for call routing.
- Options to configure the rate-limiting values, such as the expiry time, fail-count, and a list of registrars to be used for registration.

## Examples

The following example shows how to set the registrar index as 2 for the SIP registration pass-through rate-limiting:

```
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# registration passthrough static rate-limit registrar-index 2
```

The following example shows how SIP registration pass-through is configured in the voice class tenant configuration mode:

```
Router(config-class)# registration passthrough system
```

## Related Commands

Command	Description
<b>voice-class sip registration passthrough static rate-limit</b>	Sets the SIP registration pass-through rate limiting options on a dial peer.

# 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 or voice class tenant configuration mode. To reset to the default, use the **no** form of this command.

**rel1xx** {**supported** *value* | **require** *value* | **disable** | **system**}  
**no rel1xx**

## Syntax Description

<b>supported</b> <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.
<b>require</b> <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.
<b>disable</b>	Disables the use of reliable provisional responses.
<b>system</b>	Use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

## Command Default

**supported** with the 100rel value

## Command Modes

SIP configuration mode (conf-voi-serv)

Voice class tenant configuration (config-class)

Dial-peer configuration mode

## 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.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

## 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
```

The following example shows use of the **rel1xx** command in the voice class tenant configuration mode:

```
Router(config-class)# rel1xx system
```

### Related Commands

Command	Description
<b>sip</b>	Enters SIP configuration mode from voice-service VoIP configuration mode.
<b>voice-class sip rel1xx</b>	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 or voice class tenant configuration mode. To disable Remote-Party-ID translation, use the no form of this command.

**remote-party-id system**  
**no remote-party-id**

<b>Syntax Description</b>	<b>system</b>	Specifies that the SIP header Remote-Party-ID use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.
---------------------------	---------------	---

**Command Default** Remote-Party-ID translation is enabled.

**Command Modes** SIP UA configuration  
 Voice class tenant configuration (config-class)

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.2(13)T	This command was introduced.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

**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 that is 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
```

The following example shows the Remote-Party-ID translation being enabled in the voice class tenant configuration mode:

```
Router(config-class)# remote-party-id system
```



**Related Commands**

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

## remote-url

To configure the url the application that will be used by the service provider, use the **remote-url** command. The provider uses this url to authenticate and communicate with the application. To delete the configured url, use the **no** form of this command.

**remote-url** [{*url-number*}] *url*

### Syntax Description

<i>url-number</i>	(optional) URL number. Range is from 1 to 8. <b>Note</b> You can configure only one URL for XSVC service provider in secure mode.
<i>url</i>	Specifies the URL that the service provider will be using in the messages. In secure mode, only HTTPS URL can be configured. <b>Note</b> In secure mode, only IPv4 address can be configured. IPv6 address and domain name cannot be configured.

### Command Default

No default behavior or values.

### Command Modes

uc wsapi mode  
uc secure-wsapi mode

### Command History

Release	Modification
15.2(2)T	This command was introduced.
Cisco IOS XE Everest 16.6.1	This command extended support for configuring Cisco Unified Communication IOS services environment using HTTPS connection.

### Usage Guidelines

Use this command to configure the remote URL (application) that the service provider uses in messages.

### Examples

The following example configures the remote url that the the xcc service provider will use in messages while in nonsecure mode.

```
Router(config)# uc wsapi
Router(config-uc-wsapi)# provider xcc
Router(config-uc-wsapi-xcc)# no shutdown
Router(config-uc-wsapi-xcc)# remote-url 1 http://192.0.2.0:24/my_route_control
```

The following example configures the remote url that the the xcc service provider will use in messages while in secure mode.

```
Router(config)# uc secure-wsapi
Router(config-uc-wsapi)# provider xcc
Router(config-uc-wsapi-xcc)# no shutdown
Router(config-uc-wsapi-xcc)# remote-url 1 https://192.0.2.0:24/my_route_control
```

---

**Examples****Related Commands**

<b>Command</b>	<b>Description</b>
<b>provider</b>	Enables a provider service.
<b>source-address</b>	Specifies the IP address of the provider.
<b>uc wsapi</b>	Enters nonsecure Cisco Unified Communication IOS services configuration mode.
<b>uc secure-wsapi</b>	Enters secure Cisco Unified Communication IOS services configuration mode.

## ren

To configure Ring Equivalent Number of the ringer device connected to analog FXS voice port. Use the `ren <1-5>` command in voice-port configuration mode. To reset to default, use `no ren` or `ren 1`.

This command is only applicable to analog FXS voice port.

```
ren  [{ number }]  
no ren
```

---

### Command History

Syntax Description

---

### Syntax Description

<i>number</i>	REN value 1-5 for short loop analog FXS voice port. Default is 1 number REN value 1-2 for long loop analog FXS voice port. Default is 1.
---------------	--

---

### Command Default

no ren or ren 1

---

### Command Modes

Voice-port configuration

## 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

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

### Command Default

**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
<b>acc-qos</b>	Defines the acceptable QoS for any inbound and outbound call on a VoIP dial peer.

# request

To use SIP profiles to add, copy, modify, or remove Session Initiation Protocol (SIP) or Session Description Protocol (SDP) header value in a SIP request message, use the **request** command in voice class configuration mode. To disable the configuration, use the **no** form of this command.

```
request method {sdp-header | sip-header} header-name {add | copy | modify | remove} string
no request method {sdp-header | sip-header} header-name {add | copy | modify | remove} string
```

## Syntax Description

<i>method</i>	Type of message to be added, modified, or removed. It can be one of the following values: <ul style="list-style-type: none"> <li>• <b>ack</b> --SIP acknowledgment message.</li> <li>• <b>any</b> --Any SIP message.</li> <li>• <b>bye</b> --SIP BYE message.</li> <li>• <b>cancel</b> --SIP CANCEL message.</li> <li>• <b>comet</b> --SIP COMET message.</li> <li>• <b>info</b> --SIP INFO message.</li> <li>• <b>invite</b> --The first SIP INVITE message.</li> <li>• <b>notify</b> --SIP NOTIFY message.</li> <li>• <b>options</b> --SIP OPTIONS message.</li> <li>• <b>prack</b> --SIP PRACK message.</li> <li>• <b>publish</b> --SIP PUBLISH message.</li> <li>• <b>refer</b> --SIP REFER message.</li> <li>• <b>register</b> --SIP REGISTER message.</li> <li>• <b>reinvite</b> --SIP REINVITE message.</li> <li>• <b>subscribe</b> --SIP SUBSCRIBE message.</li> <li>• <b>update</b> --SIP UPDATE message.</li> </ul>
<b>sdp-header</b>	Specifies an SDP header.
<b>sip-header</b>	Specifies a SIP header.
<i>header-name</i>	SDP or SIP header name.
<b>add</b>	Adds a header.
<b>copy</b>	Copies a header.
<b>modify</b>	Modifies a header.

<b>remove</b>	Removes a header.
<i>string</i>	String to be added, copied, modified, or removed as a header.  <b>Note</b> If you use the <b>copy</b> keyword, you must provide a matching pattern followed by the variable name for the <i>string</i> argument.

**Command Default**

SIP profiles are not modified to add, copy, modify, or remove SIP or SDP header values.

**Command Modes**

Voice class configuration (config-class)

**Command History**

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

**Usage Guidelines**

If there are interoperability issues with Cisco UBE, the Cisco UBE will not work with the default SIP signaling. Hence, you must modify the SIP profiles to add, copy, modify, or remove SIP or SDP header values, and therefore enable Cisco UBE to work with SIP signaling.

Use the **request** command to modify SIP profiles for a request message. You can add, copy, modify, or remove SIP or SDP header values in an outgoing SIP request message.

**Examples**

The following example shows how to copy a SIP header value in a SIP request message:

```
Router(config)# voice class sip-profiles 10
Router(config-class)# request invite sip-header contact copy "(.*)" u01
```

**Related Commands**

Command	Description
<b>response</b>	Modifies a SIP profile to add, copy, modify, or remove a SIP or SDP header value from a SIP response message.



## request-peer-header

To use SIP profiles to copy a peer header from an outgoing Session Initiation Protocol (SIP) request message, use the **request-peer-header** command in voice class configuration mode. To disable the configuration, use the **no** form of this command.

```
request method peer-header sip {sip-req-uriheader-name} copy pattern variable
no request method peer-header sip {sip-req-uriheader-name} copy pattern variable
```

Syntax Description	
<i>method</i>	Type of message to be copied. You can specify any of the following values: <ul style="list-style-type: none"> <li>• <b>ack</b> --SIP acknowledgment message.</li> <li>• <b>any</b> --SIP message.</li> <li>• <b>bye</b> --SIP BYE message.</li> <li>• <b>cancel</b> --SIP CANCEL message.</li> <li>• <b>comet</b> --SIP COMET message.</li> <li>• <b>info</b> --SIP INFO message.</li> <li>• <b>invite</b> --First SIP INVITE message.</li> <li>• <b>notify</b> --Specifies SIP NOTIFY message.</li> <li>• <b>options</b> --SIP OPTIONS message.</li> <li>• <b>prack</b> --SIP PRACK message.</li> <li>• <b>publish</b> --SIP PUBLISH message.</li> <li>• <b>refer</b> --SIP REFER message.</li> <li>• <b>register</b> --SIP REGISTER message.</li> <li>• <b>reinvite</b> --SIP REINVITE message.</li> <li>• <b>subscribe</b> --SIP SUBSCRIBE message.</li> <li>• <b>update</b> --SIP UPDATE message.</li> </ul>
<b>sip</b>	Specifies that the SIP header must be copied from the peer call leg.
<b>sip-req-uri</b>	Specifies the SIP request Uniform Resource Identifier (URI) to be copied from the peer call leg.
<i>header-name</i>	Header name from which the values must be copied.
<b>copy</b>	Copies a header.
<i>pattern</i>	Match pattern.
<i>variable</i>	Variable to which the pattern value must be copied. The range is from u01 to u99.

**Command Default** No SIP profiles are modified to copy a peer header in an outgoing SIP request message.

**Command Modes** Voice class configuration (config-class)

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

**Usage Guidelines** If there are interoperability issues with Cisco UBE, then the Cisco UBE will not be able to work with the default SIP signaling. Hence, you must modify the SIP profiles to add, copy, modify, or remove SIP or SDP header values, and therefore enable Cisco UBE to work with SIP signaling.

Configure the **request peer-header** command to use SIP profiles to copy a peer header from an outgoing SIP request message.

**Examples** The following example shows how to copy a peer header in an outgoing SIP request message:

```
Router(config)# voice class sip-profiles 10
Router(config-class)# request invite peer-header sip contact copy "(.*)" u01
```

Command	Description
<b>response peer-header</b>	Uses SIP profiles to copy a peer header from an outgoing SIP response message.

## 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	Parameter	Description
	<b>outstanding</b>	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.
	<b>timeout</b>	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
	<b>ixi transport http</b>	Enters XML transport configuration mode.
	<b>ixi application mib</b>	Enters XML application configuration mode.
	<b>response size (XML transport)</b>	Set the XML transport fragment size.

# requi-passing

To enable pass through of the host part of the Request-URI and To SIP headers, use the **requi-passing** command in the Session Initiation Protocol (SIP) configuration mode. To disable this configuration, use the **no** form of the command.

**requi-passing**  
**no requi-passing**

**Syntax Description** This command has no keywords or arguments.

**Command Default** The outbound Request-URI is set to session target.

**Command Modes** Session Initiation Protocol (SIP) configuration mode (conf-voi-serv).  
 Voice class tenant configuration (config-class).

Command History	Release	Modification
	15.4(1)T	This command was introduced.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

**Usage Guidelines** By default, Cisco Unified Border Element sets the host part of the URI to the value configured under the session target of the outbound dial peer.

## Example

The following example shows how to enable pass through of the host part of the Request-URI and To SIP headers using the **requi-passing** command:

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# sip
Device(conf-serv-sip)# requi-passing
Device(conf-serv-sip)# end
```

## Related Commands

Command	Description
<b>contact-passing</b>	Configures pass-through of the contact header from one leg to the other leg for 302 pass-through.
<b>session target sip-uri</b>	Derives session target from incoming URI.
<b>voice-class sip requi-passing</b>	Enables the pass through of SIP URI headers.

# 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
```

## reset timer expires

To globally configure Cisco Unified Communications Manager Express (Cisco Unified CME), a Cisco IOS voice gateway, or a Cisco Unified Border Element (Cisco UBE) to reset the expires timer upon receipt of a Session Initiation Protocol (SIP) 183 Session In Progress message, use the **reset timer expires** command in voice service SIP configuration mode or voice class tenant configuration mode. To globally disable resetting of the expires timer upon receipt of SIP 183 messages, use the **no** form of this command.

**reset timer expires 183 system**  
**no reset timer expires 183 system**

### Syntax Description

<b>183</b>	Specifies resetting of the expires timer upon receipt of SIP 183 Session In Progress messages.
<b>system</b>	Specifies that the expires timer requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

### Command Default

The expires timer is not reset after receipt of SIP 183 Session In Progress messages and a session or call that is not connected within the default expiration time (three minutes) is dropped.

### Command Modes

Voice service SIP configuration (conf-serv-sip)

Voice class tenant configuration (config-class)

### Command History

Release	Modification
15.0(1)XA	This command was introduced.
15.1(1)T	This command was integrated into Cisco IOS Release 15.1(1)T.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.

### Usage Guidelines

In some scenarios, early media cut-through calls (such as emergency calls) rely on SIP 183 with session description protocol (SDP) Session In Progress messages to keep the session or call alive until receiving a FINAL SIP 200 OK message, which indicates that the call is connected. In these scenarios, the call can time out and be dropped if it does not get connected within the default expiration time (three minutes).



**Note** The expires timer default is three minutes. However, you can configure the expiration time to a maximum of 30 minutes using the **timers expires** command in SIP user agent (UA) configuration mode.

To prevent early media cut-through calls from being dropped because they reach the expires timer limit, use the **reset timer expires** command in voice service SIP configuration mode to globally enable all dial peers on Cisco Unified CME, Cisco IOS voice gateways, or Cisco UBEs to reset the expires timer upon receipt of any SIP 183 message.

To configure the reset timer expiration setting for an individual dial peer, use the **voice-class sip reset timer expires** command in dial peer voice configuration mode. To disable the expires timer reset on receipt of SIP 183 messages function, use the **no reset timer expires** command in voice service SIP configuration mode.

## Examples

The following example shows how to globally configure all dial peers on Cisco Unified CME, a Cisco IOS voice gateway, or a Cisco UBE to reset the expires timer each time a SIP 183 message is received:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# reset timer expires 183
```

The following example shows how to reset the expire timer each time a SIP 183 message is received in the voice class tenant configuration mode:

```
Router(config-class)# reset timer expires 183 system
```

## Related Commands

Command	Description
<b>timers expires</b>	Specifies how long a SIP INVITE request remains valid before it times out if no appropriate response is received for keeping the session alive.
<b>voice-class sip reset timer expires</b>	Configures an individual dial peer on Cisco Unified CME, a Cisco IOS voice gateway, or a Cisco UBE to reset the expires timer upon receipt of a SIP 183 message.

## resource (voice)

To configure parameters for monitoring resources, use the **resource** command in voice-class configuration mode. To disable the configuration for monitoring resources, use the **no** form of this command.

```
resource {cpu {1-min-avg | 5-sec-avg} | ds0 | dsp | mem {io-mem | proc-mem | total-mem}} [threshold
high threshold-value low threshold-value]
no resource {cpu | ds0 | dsp | mem}
```

### Syntax Description

<b>cpu</b>	Reports the CPU utilization information.
<b>1-min-avg</b>	Collects the CPU data for an average of one minute.
<b>5-sec-avg</b>	Collects the CPU data for an average of five seconds.
<b>ds0</b>	Reports utilization information for the DS0 port.
<b>dsp</b>	Reports utilization information for the digital signal processor (DSP) channel.
<b>mem</b>	Reports the memory utilization information.
<b>io-mem</b>	Reports the input/output memory utilization information.
<b>proc-mem</b>	Reports the process memory utilization information.
<b>total-mem</b>	Reports the complete memory utilization information.
<b>threshold</b>	Configures the high and low threshold values for the critical resources.
<b>high</b>	(Optional) Configures the resource high watermark value.
<b>low</b>	(Optional) Configures the resource low watermark value.
<i>threshold-value</i>	Threshold value, in percentage.

### Command Default

Critical gateway resources are not monitored.

### Command Modes

Voice-class configuration mode (config-class)

### Command History

Release	Modification
15.1(2)T	This command was introduced.

### Usage Guidelines

Use the **resource** command to configure parameters for critical resources such as CPU, memory, DS0, and DSP to report the utilization status to external entities using the gateway resources for call handling. You can use the **voice class resource-group** command to enter voice-class configuration mode and configure resource groups. Each resource group has a unique number that identifies a group of resources to be monitored.

When you configure the high watermark values for any of the monitoring resources, be sure not to use more resources than available on the gateway. The high and low watermark values for threshold only indicate that



the gateway might run out of resources soon. However, the gateway must still be able to trigger threshold-based reporting to the routing/monitoring entity.

When you configure the low watermark value for the threshold, be sure not to underutilize the gateway resources.

### Examples

The following example shows how to configure CPU to report the utilization information to the external entities:

```
Router> enable
Router# configure terminal
Router(config)# voice class resource-group 1
Router(config-class)# resource cpu 1-min-avg threshold high 10 low 2
```

### Related Commands

Command	Description
<b>debug rai</b>	Enables debugging for Resource Allocation Indication (RAI).
<b>periodic-report interval</b>	Configures periodic reporting parameters for gateway resource entities.
<b>rai target</b>	Configures the SIP RAI mechanism.
<b>show voice class resource-group</b>	Displays the resource group configuration information for a specific resource group or all resource groups.
<b>voice class resource-group</b>	Enters voice-class configuration mode and assigns an identification tag number for a resource group.

## 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

<b>all</b>	(Optional) High- and low-parameter settings are applied to all monitored H.323 resources. This is the default condition.
<b>high</b> <i>percentage-value</i>	(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.
<b>low</b> <i>percentage-value</i>	(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 <b>low</b> 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
<b>show call resource voice stats</b>	Displays resource statistics for an H.323 gateway.
<b>show call resource voice threshold</b>	Displays the threshold configuration settings and status for an H.323 gateway.
<b>show gateway</b>	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

<i>identifier</i>	Identifies the DSP resource to be configured. Valid values consist of alphanumeric characters, plus "_" and "-".
<b>dsps</b>	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
<b>debug mediacard</b>	Displays debugging information for DSPRM.
<b>show mediacard</b>	Displays information about the selected media card.

## response (voice)

To use SIP profiles to add, copy, modify, or remove Session Initiation Protocol (SIP) or Session Description Protocol (SDP) header value in a SIP response message, use the **response** command in voice class configuration mode. To disable the configuration, use the **no** form of this command.

```
response option {sdp-header | sip-header} header-name {add | copy | modify | remove} string
no response option {sdp-header | sip-header} header-name {add | copy | modify | remove} string
```

Syntax Description	
<i>option</i>	Response code to be added, copied, modified, or removed. You can specify one of the following values: <ul style="list-style-type: none"> <li>• <i>code</i> --Response code value. It can be one of the following values:               <ul style="list-style-type: none"> <li>• 100</li> <li>• 180 to 183</li> <li>• 200</li> <li>• 102</li> <li>• 300 to 302</li> <li>• 305</li> <li>• 380</li> <li>• 400 to 423</li> <li>• 480 to 489</li> <li>• 491</li> <li>• 493</li> <li>• 500 to 505</li> <li>• 515</li> <li>• 580</li> <li>• 600</li> <li>• 603</li> <li>• 604</li> <li>• 606</li> </ul> </li> <li>• <b>any</b> --Adds, copies, modifies, or removes any response message.</li> </ul>
<b>sdp-header</b>	Specifies SDP header.
<b>sip-header</b>	Specifies SIP header.
<i>header-name</i>	SDP or SIP header name.
<b>add</b>	Adds a header.
<b>copy</b>	Copies a header.
<b>modify</b>	Modifies a header.
<b>remove</b>	Removes a header.

<i>string</i>	String to be added as a header.
---------------	---------------------------------

**Command Default**

No SIP profile is modified to add, copy, modify, or remove a SIP header value.

**Command Modes**

Voice class configuration (config-class)

**Command History**

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

**Usage Guidelines**

If there are interoperability issues with Cisco UBE, the Cisco UBE will not be able to work with the default SIP signaling. Hence, you must modify the SIP profiles to add, copy, modify, or remove SIP header values, to enable Cisco UBE to work with SIP signaling.

Use the **response** command to modify SIP profiles for a response message. You can add, copy, modify, or remove SIP or SDP header values in an outgoing SIP response message.

**Examples**

The following example shows how to copy a SIP header value in a SIP response message:

```
Router(config)# voice class sip-profiles 10
Router(config-class)# response 409 sip-header to copy string1
```

**Related Commands**

Command	Description
<b>request</b>	Modifies a SIP profile to add, copy, modify, or remove a SIP or SDP header value from an outgoing SIP request message.

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

### Syntax Description

<b>formatted</b>	Response parameters in formatted human readable XML.
<b>timeout</b>	Application specified response timeout.
<b>-1</b>	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
<b>ixi application mib</b>	Enters XML application configuration mode.
<b>request (XML transport)</b>	Set the XML transport mode request handling parameters.

## response peer-header

To use SIP profiles to copy a peer header value in a SIP response message, use the **response peer-header** command in voice class configuration mode. To disable the configuration, use the **no** form of this command.

**response** {*code* | **any**} **peer-header sip** {*sip-req-uri**header-name*} **copy pattern variable**  
**no response option peer-header sip** {*sip-req-uri**header-name*} **copy pattern variable**

### Syntax Description

<i>code</i>	Response code to be copied. You can specify one of the following values: <ul style="list-style-type: none"> <li>• <b>100</b></li> <li>• <b>180 to 183</b></li> <li>• <b>200</b></li> <li>• <b>102</b></li> <li>• <b>300 to 302</b></li> <li>• <b>305</b></li> <li>• <b>380</b></li> <li>• <b>400 to 423</b></li> <li>• <b>480 to 489</b></li> <li>• <b>491</b></li> <li>• <b>493</b></li> <li>• <b>500 to 505</b></li> <li>• <b>515</b></li> <li>• <b>580</b></li> <li>• <b>600</b></li> <li>• <b>603</b></li> <li>• <b>604</b></li> <li>• <b>606</b></li> </ul> <ul style="list-style-type: none"> <li>• <b>any</b> --Adds, copies, modifies, or removes any response message.</li> </ul>
<b>any</b>	Adds, copies, modifies, or removes any response message.
<b>sip</b>	Specifies that the SIP header must be copied from the peer call leg.
<b>sip-req-uri</b>	Specifies the SIP request Uniform Resource Identifier (URI) to be copied from the peer call leg.
<i>header-name</i>	Header name from which the peer header values must be copied.
<b>copy</b>	Copies a header.
<i>pattern</i>	Match pattern.
<i>variable</i>	The destination variable name. The range is from u01 to u99.

### Command Default

No SIP profile is modified.



**Command Modes**

Voice class configuration (config-class)

**Command History**

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

**Usage Guidelines**

If there are interoperability issues with Cisco UBE, the Cisco UBE will not be able to work with the default SIP signaling. Hence, you must modify the SIP profiles to add, copy, modify, or remove SIP or SDP header values, to enable Cisco UBE to work with SIP signaling.

Use the **response peer-header** command to copy a peer header value in a SIP response message.

**Examples**

The following example shows how to copy a peer header value in a SIP response message:

```
Router(config)# voice class sip-profiles 10
Router(config-class)# response 200 peer-header sip contact copy "(.*) " u01
```

**Related Commands**

Command	Description
<b>request peer-header</b>	Uses SIP profiles to copy a peer header value in a SIP request message.

## 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
<b>ixi transport http</b>	Enters XML transport configuration mode.
<b>ixi application mib</b>	Enter XML application configuration mode.
<b>request (XML transport)</b>	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
<b>connection -timeout</b>	Configures the time for which a connection is maintained after completion of a communication exchange.
<b>customer -id</b>	Identifies a carrier or ISP with a settlement provider.
<b>device -id</b>	Specifies a gateway associated with a settlement provider.
<b>encryption</b>	Sets the encryption method to be negotiated with the provider.
<b>max -connection</b>	Sets the maximum number of simultaneous connections to be used for communication with a settlement provider.
<b>retry -delay</b>	Sets the time between attempts to connect with the settlement provider.
<b>retry -limit</b>	Sets the maximum number of attempts to connect to the provider.

<b>Command</b>	<b>Description</b>
<b>session -timeout</b>	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.
<b>show settlement</b>	Displays the configuration for all settlement server transactions.
<b>shutdown/no shutdown</b>	Deactivates the settlement provider/activates the settlement provider.
<b>type</b>	Configures an SAA-RTR operation type.
<b>url</b>	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
<b>auto-config</b>	Enables auto-configuration or enters auto-config application configuration mode for the SCCP application.
<b>show auto-config</b>	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 voice class tenant configuration mode. To reset to the default, use the no form of this command.

**retry bye** *number* **system**  
**no retry bye** *number* **system**

Syntax Description	
<i>number</i>	Number of BYE retries. Range is from 1 to 10. The default is 10.
<b>system</b>	Specifies that the requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 10 retries

**Command Modes** SIP UA configuration

Voice class tenant configuration (config-class)

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.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

**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

Router(config-class)# retry bye system

```

Related Commands	Command	Description
	<b>default</b>	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.
	<b>retry comet</b>	Configures the number of times that a COMET request is retransmitted to the other user agent.
	<b>retry invite</b>	Configures the number of times that a SIP INVITE request is retransmitted to the other user agent.
	<b>retry notify</b>	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
	<b>retry prack</b>	Configures the number of times that the PRACK request is retransmitted to the other user agent.
	<b>retry rel1xx</b>	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
	<b>retry response</b>	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
	<b>sip-ua</b>	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 or voice class tenant configuration mode. To reset to the default, use the **no** form of this command.

**retry cancel** *number* **system**

**no retry cancel** *number* **system**

Syntax Description	
<i>number</i>	Number of CANCEL retries. Range is from 1 to 10. Default is 10.
<b>system</b>	Specifies that the cancel requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 10 retries

**Command Modes** SIP UA configuration

Voice class tenant configuration (config-class)

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.
	15.6(2)T	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

**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
```



The following example sets the number of cancel retries in the voice class tenant configuration mode:

```
Router(config-class)# retry cancel system
```

#### Related Commands

Command	Description
<b>default</b>	Resets the value of a command to its default.
<b>retry bye</b>	Configures the number of times that a BYE request is retransmitted to the other user agent.
<b>retry comet</b>	Configures the number of times that a COMET request is retransmitted to the other user agent.
<b>retry invite</b>	Configures the number of times that a SIP INVITE request is retransmitted to the other user agent.
<b>retry notify</b>	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
<b>retry prack</b>	Configures the number of times that the PRACK request is retransmitted to the other user agent.
<b>retry rel1xx</b>	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
<b>retry response</b>	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
<b>sip-ua</b>	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**

<b>Syntax Description</b>	<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

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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
```

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	retry bye	Configures the number of times that a BYE request is retransmitted to the other user agent.
	retry cancel	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
	retry invite	Configures the number of times that a SIP INVITE request is retransmitted to the other user agent.

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

# retry info

To configure the number of times, that an INFO request is retransmitted to the other user agent, use **retry info** command in SIP UA configuration mode or voice class tenant configuration mode.

**retry info** *number* [**system**]

**no retry update**

<b>Syntax Description</b>	<i>number</i> Number of INFO retries. Range is from 1 to 10. Default is 6.
	<b>system</b> Specifies that the INFO requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 6 retries

**Command Modes** SIP UA configuration  
Voice class tenant configuration

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	Cisco IOS 15.6(2)T and Cisco IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

**Usage Guidelines** Configures the number of times, that an INFO request is retransmitted to the other user agent.

## Example

In sip-ua mode:

```
Device> enable
Device# configure terminal
Device(config)# sip-ua
Device(config-sip-ua)# retry info 8
```

In voice class tenant mode:

```
Device> enable
Device# configure terminal
Device(config)# voice class tenant 1
Device(config-class)# retry info 8
```

# 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**

<b>Syntax Description</b>	<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

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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
```

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>access-policy</b>	Requires that a neighbor be explicitly configured.
	<b>inbound ttl</b>	Sets the inbound time-to-live value.
	<b>outbound retry-interval</b>	Defines the retry period for attempting to establish the outbound relationship between border elements.
	<b>retry window</b>	Defines the total time for which a border element attempts delivery.
	<b>service-relationship</b>	Establishes a service relationship between two border elements.
	<b>shutdown</b>	Enables or disables the border element.
	<b>usage-indication</b>	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 or voice class tenant configuration mode. To reset to the default, use the **no** form of this command.

**retry invite** *number* **system**  
**no retry invite** *number* **system**

<b>Syntax Description</b>	<i>number</i>	Number of INVITE retries. Range is from 1 to 10. Default is 6.
	<b>system</b>	Specifies that the INVITE requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 6 retries

**Command Modes** SIP UA configuration  
 Voice class tenant configuration (config-class)

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

**Usage Guidelines** To reset this command to the default value, you can also use the **default** command.  
 When using the dial-peer rotary function, ensure that the **retry invite** command value is set to 4 or less.



**Note** CUBE uses the exponential backoff series algorithm (1, 2, 4, 8, 16, 32, 64, 128, ... seconds) to retry the invites. The invites are resent after each exponential delay. For example, if the retry-invite value is 6 (default), then the CUBE uses 6 exponential backoff elements and resend invite after each exponential delay (that is, re-sends invite after 1, 2, 4, 8, 16, 32, seconds). In this case, the final invite is sent after 64 seconds ( $1+2+4+8+16+32=64$ ). If you reset the retry value to 2, then the CUBE uses 2 exponential backoff elements (that is, re-sends invite after 1, 2 seconds). In this case, the final invite is sent after 3 seconds ( $1+2=3$ ).

## Examples

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

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

The following example sets the number of invite retries to 2 for tenant 1 in the voice class tenant configuration mode:

```
Router> enable
Router# configure terminal
Router(config)# voice class tenant 1
Router(config-class)# retry invite 2
```

## Related Commands

Command	Description
<b>default</b>	Resets the value of a command to its default.
<b>retry bye</b>	Configures the number of times that a BYE request is retransmitted to the other user agent.
<b>retry cancel</b>	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
<b>retry comet</b>	Configures the number of times that a COMET request is retransmitted to the other user agent.
<b>retry notify</b>	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
<b>retry prack</b>	Configures the number of times that the PRACK request is retransmitted to the other user agent.
<b>retry rel1xx</b>	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
<b>retry response</b>	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
<b>sip-ua</b>	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.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

## 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
<b>busyout monitor keepalive</b>	Selects a voice port or ports to be busied out in cases of a keepalive failure.
<b>keepalive target</b>	Identifies a SIP server that will receive keepalive packets from the SIP gateway.
<b>keepalive trigger</b>	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.
<b>timers keepalive</b>	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 or voice class tenant configuration mode. To reset to the default, use the **no** form of this command.

**retry notify** *number* **system**

**no retry notify system**

Syntax Description	
<i>number</i>	Number of notify message retries. Range is from 1 to 10. Default is 10.
<b>system</b>	Specifies that the notify messages use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 10 retries

**Command Modes** SIP UA configuration

Voice class tenant configuration (config-class)

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.
	Cisco IOS XE Release 2.5	This command was integrated into Cisco IOS XE Release 2.5.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

**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
```

The following example configures a notify message to be retransmitted in the voice class tenant configuration mode:

```
Router(config-class)# retry notify system
```

## Related Commands

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

# retry options

To configure the number of times, that an OPTIONS request is retransmitted to the other user agent, use **retry options** command in SIP UA configuration mode or voice class tenant configuration mode.

**retry options** *number* [**system**]

**no retry options**

<b>Syntax Description</b>	<i>number</i>	Number of OPTIONS retries. Range is from 1 to 10. Default is 6.
	<b>system</b>	Specifies that the OPTIONS requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.
<b>Command Default</b>	6 retries	
<b>Command Modes</b>	SIP UA configuration	
	Voice class tenant configuration	
<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	Cisco IOS 15.6(2)T and Cisco IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.
<b>Usage Guidelines</b>	Configures the number of times, that an OPTIONS request is retransmitted to the other user agent.	

## Example

In sip-ua mode:

```
Device> enable
Device# configure terminal
Device(config)# sip-ua
Device(config-sip-ua)# retry options 8
```

In voice class tenant mode:

```
Device> enable
Device# configure terminal
Device(config)# voice class tenant 1
Device(config-class)# retry options 8
```

# 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 or voice class tenant configuration mode. To reset to the default, use the **no** form of this command.

**retry prack** *number* **system**  
**no** **retry prack** **system**

Syntax Description	
<b>number</b>	Number of PRACK retries. Range is from 1 to 10. Default is 10.
<b>system</b>	Specifies that the prack requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 10 retries

**Command Modes** SIP UA configuration

Voice class tenant configuration (config-class)

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.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

**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
```

The following example configures a PRACK request to be retransmitted in the voice class tenant configuration mode:

```
Router(config-class)# retry prack system
```

#### Related Commands

Command	Description
<b>retry bye</b>	Configures the number of times that a BYE request is retransmitted to the other user agent.
<b>retry cancel</b>	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
<b>retry comet</b>	Configures the number of times that a COMET request is retransmitted to the other user agent.
<b>retry invite</b>	Configures the number of times that a SIP INVITE request is retransmitted to the other user agent.
<b>retry notify</b>	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
<b>retry rel1xx</b>	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
<b>retry response</b>	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
<b>show sip-ua retry</b>	Displays the SIP retry attempts.
<b>show sip-ua statistics</b>	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 or voice class tenant configuration mode. To reset to the default, use the **no** form of this command.

**retry refer** *number* **system**

**no** **retry refer** **system**

Syntax Description	
<i>number</i>	Number of Refer request retries. Range is from 1 to 10. Default is 10.
<b>system</b>	Specifies that the REFER requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 10 retries

**Command Modes** SIP UA configuration

Voice class tenant configuration (config-class)

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.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

**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
```

The following example configures a Refer request to be retransmitted in the voice class tenant configuration mode:

```
Router(config-class)# retry refer system
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>show sip-ua retry</b>	Displays the SIP retry attempts.
<b>show sip-ua statistics</b>	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 user-agent configuration mode or voice class tenant configuration mode. To reset this number to the default, use the **no** form of this command.

**retry register** *retries* **system**[**exhausted-random-interval** **minimum** *minutes* **maximum** *minutes*]  
**no** **retry register**

## Syntax Description

<i>retries</i>	Total number of register messages that the gateway should send. The range is from 1 to 10. The default is 6 retries.
<b>exhausted-random-interval</b>	Specifies the register request to be generated within the defined range of time intervals.
<b>minimum</b> <i>minutes</i>	Specifies the minimum time interval range, in minutes, that will be used as the interval before the next registration is sent.
<b>maximum</b> <i>minutes</i>	Specifies the maximum time interval range, in minutes, that will be used as the interval before the next registration is sent.
<b>system</b>	Specifies that the register messages use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

## Command Default

The gateway sends 6 retries.

## Command Modes

SIP UA configuration (config-sip-ua)

Voice class tenant configuration (config-class)

## 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(22)T	This command was modified. Support for IPv6 was added.
12.4(22)YB	This command was modified. The <b>exhausted-random-interval</b> keyword was added.
15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.



**Usage Guidelines**

Use the default number when possible. Lower values such as 1 may lead to the message not being received by the other user agent.

**Examples**

The following example shows how to configure the gateway to send 9 register messages:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# retry register 9
```

The following example shows how to configure the gateway to send 6 register messages and choose a random number between 2 and 5 as the interval before sending the next registration message:

```
Router> enable
Router# configure terminal
Router(config)# sip-ua
Router(config-sip-ua)# retry register 6 exhausted-random-interval minimum 2 maximum 5
```

The following example configures the gateway to register messages in the voice class tenant configuration mode:

```
Router(config-class)# retry register system
```

**Related Commands**

Command	Description
<b>registrar</b>	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.
<b>timers register</b>	Sets how long the SIP user agent 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 or voice class tenant configuration mode. To reset to the default, use the **no** form of this command.

**retry rel1xx** *number* **system**  
**no** **retry rel1xx** **system**

## Syntax Description

<i>number</i>	Number of reliable 1xx retries. Range is from 1 to 10. Default is 6.
<b>system</b>	Specifies that the reliable 1xx response is retransmitted use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

## Command Default

6 retries

## Command Modes

SIP UA configuration

Voice class tenant configuration (config-class)

## 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.
15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

## 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
```

The following example configures the reliable 1xx response to be retransmitted in the voice class tenant configuration mode:

```
Router(config-class)# retry rel1xx system
```

#### Related Commands

Command	Description
<b>retry bye</b>	Configures the number of times that a BYE request is retransmitted to the other user agent.
<b>retry cancel</b>	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
<b>retry comet</b>	Configures the number of times that a COMET request is retransmitted to the other user agent.
<b>retry invite</b>	Configures the number of times that a SIP INVITE request is retransmitted to the other user agent.
<b>retry notify</b>	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
<b>retry prack</b>	Configures the number of times the PRACK request is retransmitted.
<b>retry response</b>	Configures the number of times that the RESPONSE message is retransmitted to the other user agent.
<b>show sip-ua retry</b>	Displays the SIP retry attempts.
<b>show sip-ua statistics</b>	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 or voice class tenant configuration mode. To reset to the default, use the no form of this command.

**retry response** *number* **system**

**no retry response** **system**

Syntax Description	
<i>number</i>	Number of response retries. Range is from 1 to 10. Default is 6.
<b>system</b>	Specifies that the response messages use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 6 retries

**Command Modes** SIP UA configuration

Voice class tenant configuration (config-class)

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.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

**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

```

The following example sets the number of response retries in the voice class tenant configuration mode:

```

Router(config-class)# retry response system

```

#### Related Commands

Command	Description
<b>default</b>	Resets the value of a command to its default.
<b>retry bye</b>	Configures the number of times that a BYE request is retransmitted to the other user agent.
<b>retry cancel</b>	Configures the number of times that a CANCEL request is retransmitted to the other user agent.
<b>retry comet</b>	Configures the number of times that a COMET request is retransmitted to the other user agent.
<b>retry invite</b>	Configures the number of times that a SIP INVITE request is retransmitted to the other user agent.
<b>retry notify</b>	Configures the number of times that the Notify message is retransmitted to the user agent that initiated the transfer or Refer request.
<b>retry prack</b>	Configures the number of times the PRACK request is retransmitted.
<b>retry rel1xx</b>	Configures the number of times that the reliable 1xx response is retransmitted to the other user agent.
<b>sip-ua</b>	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 or voice class tenant configuration mode. To reset to the default, use the no form of this command.

**retry subscribe** *number* **system**  
**no** **retry subscribe** *number* **system**

<b>Syntax Description</b>	<i>number</i>	Number of SUBSCRIBE retries. Range is 1 to 10. Default is 10.
	<b>system</b>	Specifies that the SIP SUBSCRIBE message retransmitted use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 10 retries

**Command Modes** SIP UA configuration  
 Voice class tenant configuration (config-class)

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.3(4)T	This command was introduced.
	15.6(2)T and IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

**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
    
```

The following example sets the number of subscribe retries in the voice class tenant configuration mode:

```

Router(config-class)# retry subscribe system
    
```

**Related Commands**

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

# retry update

To configure the number of times, that an UPDATE request is retransmitted to the other user agent, use **retry update** command in SIP UA configuration mode or voice class tenant configuration mode.

**retry update** *number* [**system**]

**no retry update**

<b>Syntax Description</b>	<i>number</i> Number of UPDATE retries. Range is from 1 to 10. Default is 6.
	<b>system</b> Specifies that the UPDATE requests use the global sip-ua value. This keyword is available only for the tenant mode to allow it to fallback to the global configurations.

**Command Default** 6 retries

**Command Modes** SIP UA configuration  
Voice class tenant configuration

Command History	Release	Modification
	Cisco IOS 15.6(2)T and Cisco IOS XE Denali 16.3.1	This command was modified to include the keyword: <b>system</b> . This command is now available under voice class tenants.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.
	Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models under voice class tenant configuration.

**Usage Guidelines** Configures the number of times, that an UPDATE request is retransmitted to the other user agent.

### Example

In sip-ua mode:

```
Device> enable
Device# configure terminal
Device(config)# sip-ua
Device(config-sip-ua)# retry update 8
```

In voice class tenant mode:

```
Device> enable
Device# configure terminal
Device(config)# voice class tenant 1
Device(config-class)# retry update 8
```



# 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**

<b>Syntax Description</b>	<i>window -value</i> Window value, in minutes. Range is from 1 to 65535. Default is 1440 minutes (24 hours).
---------------------------	--

**Command Default** 1440 minutes (24 hours)

**Command Modes** Annex G neighbor usage configuration

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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
```

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

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

<b>Syntax Description</b>	<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

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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
```

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

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

<b>Syntax Description</b>	<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

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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
```

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

Command	Description
<b>session -timeout</b>	Sets the interval for closing the connection when there is no input or output traffic.
<b>settlement</b>	Enters settlement mode and specifies the attributes specific to a settlement provider.
<b>show settlement</b>	Displays the configuration for all settlement server transactions.
<b>shutdown</b>	Brings up the settlement provider.
<b>type</b>	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

<i>cadence-number</i>	Number that determines the ringing cadence. Range is from 0 to 2: <ul style="list-style-type: none"> <li>• Type 0 is a primary ringing cadence--default ringing cadence for the country your router is in.</li> <li>• Type 1 is a distinctive ring--0.8 seconds on, 0.4 seconds off, 0.8 seconds on, 0.4 seconds off.</li> <li>• 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.</li> </ul>
-----------------------	--

## Command Default

### Command Modes

Interface configuration

## Command History

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

## Usage Guidelines

This command applies to Cisco 800 series routers.

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

## Examples

The following example specifies the type 1 distinctive ring :

```
ring 1
```

## Related Commands

Command	Description
<b>destination -pattern</b>	Specifies the prefix, the full E.164 telephone number, or an ISDN directory number to be used for a dial peer.
<b>dial -peer voice</b>	Enters dial-peer configuration mode, defines the type of dial peer, and defines the tag number associated with a dial peer.
<b>no call -waiting</b>	Disables call waiting.
<b>port (dial -peer)</b>	Enables an interface on a PA-4R-DTR port adapter to operate as a concentrator port.

Command	Description
<b>pots distinctive -ring-guard-time</b>	Specifies a delay during which a telephone port can be rung after a previous call is disconnected (for Cisco 800 series routers).
<b>show dial -peer voice</b>	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
```

```
{ring cadence external patternXX | define}
```

```
{ring cadence patternXX | define}
```

### 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"> <li>• <b>pattern01</b> -- 2 seconds on, 4 seconds off</li> <li>• <b>pattern02</b> -- 1 second on, 4 seconds off</li> <li>• <b>pattern03</b> -- 1.5 seconds on, 3.5 seconds off</li> <li>• <b>pattern04</b> -- 1 second on, 2 seconds off</li> <li>• <b>pattern05</b> -- 1 second on, 5 seconds off</li> <li>• <b>pattern06</b> -- 1 second on, 3 seconds off</li> <li>• <b>pattern07</b> -- 0.8 second on, 3.2 seconds off</li> <li>• <b>pattern08</b> -- 1.5 seconds on, 3 seconds off</li> <li>• <b>pattern09</b> -- 1.2 seconds on, 3.7 seconds off</li> <li>• <b>pattern09</b> -- 1.2 seconds on, 4.7 seconds off</li> <li>• <b>pattern11</b> -- 0.4 second on, 0.2 second off, 0.4 second on, 2 seconds off</li> <li>• <b>pattern12</b> -- 0.4 second on, 0.2 second off, 0.4 second on, 2.6 seconds off</li> </ul>
<b>define</b>	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.
<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 <b>patternXX</b> keyword was added.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
15.0(1)M	This command was modified. The <b>external</b> keyword was added to specify the ring pattern of external calls.

**Usage Guidelines**

To specify the ring pattern for external calls, use the **ring cadence external** command. It is supported only in STCAPP. To specify the ring cadence for internal calls, use the existing **ring cadence** command. The syntax for the ring cadence external command is the same as for the **ring cadence** command.

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
<b>cptone</b>	Specifies the default tone, ring, and cadence settings according to country.
<b>ring frequency</b>	Specifies the ring frequency for a specified FXS voice port.

## ring dc-offset

To configure ring voltage threshold to prevent the ringer devices from sounding so as to ignore the lower voltages that can be produced when dialing. An increase in the ring voltage threshold value can overcome this. Use the ring dc-offset command in voice-port configuration mode. To reset to default, use the no form of this command.

This command is only applicable to analog FXS voice port with loop-length long configured.

**ring dc-offset** *volt-value*

**no ring dc-offset**

### Syntax Description

<i>volt-value</i>	volt-value 10-volts - Ring DC offset 10 volts 20-volts - Ring DC offset 20 volts 24-volts - Ring DC offset 24 volts 30-volts - Ring DC offset 30 volts 35-volts - Ring DC offset 35 volts
-------------------	--

### Command Default

no ring dc-offset

### Command Modes

Voice-port configuration

# 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"> <li>• Cisco 3600 series: 25 and 50. Default is 25.</li> </ul>
--------------	---

## Command Default

Cisco 3600 series routers: 25 Hz

## Command Modes

Voice-port configuration

## Command History

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

## Usage Guidelines

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

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

## Examples

The following example sets the ring frequency on the voice port to 25 Hz:

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

## Related Commands

Command	Description
<b>ring cadence</b>	Specifies the ring cadence for an FXS voice port.
<b>ring number</b>	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

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

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

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

## Examples

The following example 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
<b>ring frequency</b>	Specifies the ring frequency for a specified FXS voice port.

# ringing-timeout

To define the timeout period for the SCCP telephony control (STC) application feature call back, use the **ringing-timeout** command in STC application feature callback configuration mode. To return to the default timeout period, use the **no** form of this command.

**ringing-timeout** *seconds*  
**no ringing-timeout**

## Syntax Description

<i>seconds</i>	Period of time in seconds. Range: 5 to 60. Default: 30.
----------------	---

## Command Default

The default is 30 seconds.

## Command Modes

STC application feature callback configuration (config-stcapp-callback)

## Command History

Release	Modification
12.4(20)YA	This command was introduced.
12.4(22)T	This command was integrated into Cisco IOS Release 12.4(22)T.

## Usage Guidelines

This command changes the timeout period of the ringing timer from the default of 30 seconds to the specified value.

The ringing timer specifies the number of seconds during which the calling device that is in a Callback on Busy condition can receive a Callback Ringing and after which, if the calling device does not answer, the CallBack on Busy condition is cancelled.

## Examples

The following example shows how to change the timeout period of the ringing timer for CallBack on Busy from the default (30) to a new value (45).

```
Router(config)# stcapp feature callback
Router(config-stcapp-callback)# ringing-timer 45
Router(config-stcapp-callback)#
```

## Related Commands

Command	Description
<b>activation-code</b>	Defines the callback activation key sequence for CallBack on Busy.

## 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
	<b>roaming (settlement)</b>	Enables the roaming capability for a settlement provider.
	<b>settle-call</b>	Limits the dial peer to using only the specific clearinghouse identified by the specified <i>&gt;provider -&gt;number</i> .
	<b>settlement roam-pattern</b>	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
	<b>roaming (dial-peer mode)</b>	Enables the roaming capability for the dial peer.
	<b>settle-call</b>	Limits the dial peer to using only the specific clearinghouse identified by the specified <i>&gt;provider -&gt;number</i> .
	<b>settlement roam-pattern</b>	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

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
```

Command	Description
<b>ras rrq dynamic prefixes</b>	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.



**Note** When RSVP is not configured for call signaling on the Cisco UBE, use the **show dial-peer voice** command to verify the QoS settings that the signaling and media packets will be marked with. Fields corresponding to QoS negotiation in the output produced by the **show sip-ua calls** command should be ignored.

```
Local QoS Strength : BestEffort
Negotiated QoS Strength : BestEffort
Negotiated QoS Direction : None
```

## 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
<b>codec (DSP Farm profile)</b>	Specifies the codecs supported by a DSP farm profile.
<b>debug call rsvp-sync events</b>	Displays events that occur during RSVP setup.

Command	Description
<b>dspfarm profile</b>	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.
<b>show sccp connections rsvp</b>	Displays information about active SCCP connections that use RSVP.

# rtcp keepalive

To configure RTP Control Protocol (RTCP) keepalive report generation and generate RTCP keepalive packets, use the **rtcp keepalive** command in voice service configuration mode. To disable the configuration, use the **no** form of this command.

**rtcp keepalive**  
**no rtcp keepalive**

**Syntax Description** This command has no arguments or keywords.

**Command Default** The command is disabled by default.

**Command Modes** Voice service configuration (config)

Release	Modification
15.1(2)T	This command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

**Usage Guidelines** Use this command to configure RTCP keepalive report generation and generate RTCP keepalive packets. The **no** form of the command restores the default behavior.

**Examples** The following example shows how to configure RTCP keepalive report generation and generate RTCP keepalive packets:

```
Router> enable
Router# configure terminal
Router(config) voice service voip
Router(conf-voi-serv) # rtcp keepalive
```

Command	Description
<b>debug voip rtcp</b>	Enables debugging for RTCP packets.
<b>debug voip rtp</b>	Enables debugging for RTP packets.
<b>debug ip rtp protocol</b>	Enables debugging for RTP protocol.
<b>ip rtcp report interval</b>	Configures the average reporting interval between subsequent RTCP report transmissions.

## rtcp all-pass-through

To pass through all the RTCP packets in datapath. To disable the configuration, use the **no** form of this command.

```
rtcp all-pass-through
no rtcp all-passthrough
```

---

**Syntax Description** This command has no arguments or keywords.

---

**Command Default** The command is disabled by default.

---

**Command Modes** Voice service configuration (config)

---

Command History	Release	Modification
	15.1(2)T	This command was introduced.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

```
Device(conf-voi-serv)# rtcp all-pass-through
```

# rtp-media-loop count

To configure the number of media loops before Real-Time Transport Protocol (RTP) voice and video media packets are dropped, use the **rtp-media-loop count** command in voice service configuration mode. To remove this configuration, use the **no** form of this command.

**rtp-media-loop count** *number*  
**no rtp-media-loop count**

<b>Syntax Description</b>	<i>number</i> Number of media loops. The range is from 6 to 21.
---------------------------	---

<b>Command Default</b>	The number of media loops is not configured, and a default value of 6 is applied.
------------------------	---

<b>Command Modes</b>	Voice service configuration (conf-voi-serv)
----------------------	---

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	15.2(2)T3	This command was introduced.
	Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

<b>Usage Guidelines</b>	Use the <b>rtp-media-loop count</b> command when you want to control the maximum number of media loops before the RTP media packets are dropped for IP-to-IP calls. The recommended configuration is to use the default loop count of 6.
-------------------------	--

## Example

The following example shows how to configure the loop count before RTP media packets are dropped:

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# rtp-media-loop count 10
```

## 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-aacld number |
cisco-codec-fax-ack number | cisco-codec-fax-ind number | cisco-codec-gsmamrnb number |
cisco-codec-ilbc number | cisco-codec-isac 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 | lmr-tone number | nse number |
nse number | nse-tone number | opus 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 |
lmr-tone number | nse number | nse number | nse-tone number | opus number } [comfort-noise {13
| 19}]
```

### Syntax Description

<b>cisco-cas-payload</b> <i>number</i>	Cisco channel-associated signaling (CAS) RTP payload. Range: 96–127. Default: 123.
<b>cisco-clear-channel</b> <i>number</i>	Cisco clear-channel RTP payload. Range: 96–127. Default: 125.
<b>cisco-codec-aacld</b> <i>number</i>	Cisco MPEG-4 Advanced Audio Codec - Low Delay (AAC-LD) codec. Range: 96–127. Default: 114.
<b>cisco-codec-fax-ack</b> <i>number</i>	Cisco codec fax acknowledge. Range: 96–127. Default: 97.
<b>cisco-codec-fax-ind</b> <i>number</i>	Cisco codec fax indication. Range: 96–127. Default: 96.
<b>cisco-codec-gsmamrnb</b> <i>number</i>	Cisco Global System for Mobile Adaptive Multi-Rate narrowband (GSMAMR-NB) codec. Range: 96–127. Default: 117.
<b>cisco-codec-ilbc</b> <i>number</i>	Cisco Internet Low Bitrate Codec (iLBC) codec. Range: 96–127. Default: 116.
<b>cisco-codec-isac</b> <i>number</i>	Cisco internet Speech Audio Codec (iSAC) codec. Range: 96–127. Default: 124.
<b>cisco-codec-video-h263+</b> <i>number</i>	RTP video codec H.263+ payload type. Range: 96–127. Default: 118.
<b>cisco-codec-video-h264</b> <i>number</i>	RTP video codec H.264 payload type. Range: 96–127. Default: 119.
<b>cisco-fax-relay</b> <i>number</i>	Cisco fax relay. Range: 96–127. Default: 122.
<b>cisco-pcm-switch-over-alaw</b> <i>number</i>	Cisco RTP pulse code modulation (PCM) codec switch over indication (a-law). Default: 8.
<b>cisco-pcm-switch-over-ulaw</b> <i>number</i>	Cisco RTP PCM codec switch over indication (mu-law). Default: 0.

<b>cisco-rtp-dtmf-relay</b> <i>number</i>	Cisco RTP dual-tone multifrequency (DTMF) relay. Range: 96–127. Default: 121.
<b>lmr-tone</b> <i>number</i>	LMR payload type. Range: 96–127. Default: 0. The default value is set by the <b>no rtp payload-type lmr-tone</b> command.
<b>nse</b> <i>number</i>	A Named Signaling Event (NSE). Range: 96–117. Default: 100.
<b>nte</b> <i>number</i>	A named phone event (NTE). Range: 96–127. Default: 101.
<b>nte-tone</b> <i>number</i>	RFC-2833 tone payload type. Range 96–127. Default: 101.
<b>comfort-noise</b> <b>13</b> <b>19</b>	(Optional) RTP payload type of comfort noise. The July 2001 draft entitled <i>RTP Payload for Comfort Noise</i> , from the IETF (IETF) Audio or 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 <i>RTP Payload for Comfort Noise</i> draft, use 13. Use 19 only if you are connecting to older Cisco gateways that use DSPware before version 3.4.32.  <b>Note</b> 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.
<b>opus</b> <i>number</i>	Interactive speech and audio codec (opus). Range: 96–127. Default: 114.

**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	This command was modified. The <b>nte</b> and <b>comfort - noise</b> 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	This command was modified. The <b>cisco-codec-gsmamrnb</b> keyword was added.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.



Release	Modification
12.4(11)T	This command was modified. The <b>cisco-codec-ilbc</b> , <b>cisco-codec-video-h263+</b> , and <b>cisco-codec-video-h264</b> keywords were added.
12.4(15)XY	This command was modified. The <b>lmr-tone</b> and <b>nte-tone</b> keywords were added.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.
IOS Release XE 2.5	This command was integrated into Cisco IOS XE Release 2.5.
15.1(1)T	This command was modified. The <b>cisco-codec-isac</b> keyword was added.
Cisco IOS XE Amsterdam 17.3.1a	This command was modified. The <b>opus</b> keyword was added.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

## Usage Guidelines



**Note** **rtp payload** YANG configuration is supported for audio and video codecs. Other configurations that are related to fax, NSE, pcm-switchover are not supported. For example, **cisco fax-relay**, **cisco-pcm-switch-over-alaw**, **cisco-codec-fax-ack/ind**, **nse**, and **g726r16**.

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 hardcoded with Cisco proprietary meanings. Do not use the following numbers, which have preassigned values: 96, 97, 100, 117, 121–123, and 125–127.

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 shows how to identify the RTP payload type as GSMAMR-NB115:

```
Router(config-dial-peer)# rtp payload-type cisco-codec-gsmamrnb 115
```

The following example shows how to identify the RTP payload type as NTE 99:

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

The following example shows how to identify the RTP payload type for the iLBC as 100:

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

The following example shows how to identify the RTP payload type as Opus:

```
Router(config-dial-peer)# rtp payload-type opus 126
```

#### Related Commands

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

# rtp-port

To configure real-time protocol range.

**rtp-port range** *min-port max-port*

## Syntax Description

<b>min port</b>	Minimum port number.
<b>max port</b>	Maximum port number.

## Command Default

Default range of 8000–48189 is configured by default.

## Command Modes

Global configuration voice service VoIP (conf-voi-serv).

## Command History

Release	Modification
Cisco IOS XE 3.11S	The command was introduced.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

## Usage Guidelines

Configure rtp-port range to restrict the RTP ports that are used for setting up the VOIP calls on CUBE. The default global RTP port range is 8000–48189. With extended keyword, the range can be 5500–65498.

## Examples

```
Router#config t
Enter configuration commands, one per line. End with CNTL/Z.
Router(config)#voice service voip
Router(conf-voi-serv)#rtp-port
Router(conf-voi-serv)#rtp-port ?
    range port range

Router(conf-voi-serv)#rtp-port ran
Router(conf-voi-serv)#rtp-port range ?
    <8000-48198> minimum port number
    extended extended ports

Router(conf-voi-serv)#rtp-port range 8000 ?
    <8000-48198> maximum port number

Router(conf-voi-serv)#rtp-port range 8000 8012 ?
    <cr> <cr>

Router(conf-voi-serv)#rtp-port range 8000 8012
Router(conf-voi-serv)#rtp-port range exte
Router(conf-voi-serv)#rtp-port range extended ?
    <5500-65498> minimum port number extended range

Router(conf-voi-serv)#rtp-port range extended 5510 5512
Warning: Using extended port range of 5510 to 5512 could result in some tcp/udp services
not working properly due to port usage conflicts. Use caution in choosing range.
Router(conf-voi-serv)#
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>allow-connections sip to sip</b>	To allow sip-to-sip connections under voice service VoIP configuration mode for CUBE.
<b>media-address range</b>	To configure the media-address range, which enables the media gateway to allocate the available free port for a given IP address within the address range.

## rtp send-recv

To configure a Cisco IOS Session Initiation Protocol (SIP) gateway to establish a bidirectional voice path as soon as it receives a SIP 183 PROGRESS message with Session Description Protocol (SDP), use the **rtp send-recv** command in voice service SIP configuration mode. To configure the gateway to establish a backward-only media cut-through voice path upon receipt of a 183 PROGRESS message with SDP that persists until the call progresses to the connect state, use the **no** form of this command.

```
rtp send-recv
no rtp send-recv
```

**Syntax Description** This command has no arguments or keywords.

**Command Default** A bidirectional voice path is established upon receipt of a 183 PROGRESS message with SDP.

**Command Modes** Voice service SIP configuration (conf-serv-sip)

Release	Modification
12.4(15)XZ	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.

**Usage Guidelines** The default behavior on a Cisco IOS SIP gateway is to establish a bidirectional voice path from the moment it receives a SIP 183 PROGRESS message with SDP. However, this can result in clipping on some voice platforms if both parties send audio at the same time, such as during a call setup process when interactive voice response (IVR) and a caller both speak simultaneously. To establish the voice path in the backward direction only until the call is connected, use the **no rtp send-recv** command in voice service SIP configuration mode.

A backward-only voice path operates only during the connection attempt--once a call is connected, the voice path automatically converts to bidirectional sending and receiving of Real-Time Transport Protocol (RTP) packets and RTP control packets (RTCPs). However, if the **no rtp send-recv** command is configured on a SIP gateway, no inband or RFC 2833-based dual tone multifrequency (DTMF) digits can be sent in the forward direction until after the call is connected and the bidirectional voice path is established.

### Examples

The following example enables RTP backward-only media cut-through on a Cisco IOS SIP gateway:

```
Router> enable
Router# configure terminal
Router(config)# voice service voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# no rtp send-recv
```

## rtp-ssrc multiplex

To multiplex Real-Time Transport Control Protocol (RTCP) packets with RTP packets and to send multiple synchronization source in RTP headers (SSRCs) in a RTP session, use the **rtp-ssrc multiplex** command in voice service or dial peer voice configuration mode. To disable the configuration, use the **no** form of this command.

### Syntax Available Under Voice Service Configuration Mode

**rtp-ssrc multiplex**

**no rtp-ssrc multiplex**

### Syntax Available Under Dial Peer Voice Configuration Mode

**rtp-ssrc multiplex [system]**

**no rtp-ssrc multiplex [system]**

#### Syntax Description

<b>system</b>	Uses the system value. This is the default value.
---------------	---

#### Command Default

Under voice service configuration mode, the **rtp-ssrc multiplex** command is not enabled and hence there is no interoperation with Cisco TelePresence System (CTS).

At the dial-peer level, the **rtp-ssrc multiplex** command uses the global configuration level settings.

#### Command Modes

Voice service configuration (conf-voi-serv)

Dial peer voice configuration (config-dial-peer)

#### Command History

Release	Modification
12.4(15)XY	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.
Cisco IOS XE Cupertino 17.7.1a	Introduced support for YANG models.

#### Usage Guidelines

The **rtp-ssrc multiplex** command is used for the interoperation with CTS.

#### Examples

The following example shows how to multiplex RTCP packets with RTP packets and send multiple SSRCs in a RTP session:

```
Router# configure terminal
Router(config)# dial-peer voice 234 voip
Router(config-dial-peer)# rtp-ssrc multiplex system
```

## 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**

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

<b>Command Default</b>	10 minutes
------------------------	------------

<b>Command Modes</b>	Global configuration
----------------------	----------------------

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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
```

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

Command	Description
show rtsp client session	Displays cumulative information about the RTSP session records.



## rtsp client rtpsetup enable

To configure a router to send the IP address in a Real Time Streaming Protocol (RTSP) setup message, use the **rtsp client rtpsetup enable** command in global configuration mode. To disable the configuration, use the **no** form of this command.

```
rtsp client rtpsetup enable
no rtsp client rtpsetup enable
```

**Syntax Description** This command has no arguments or keywords.

**Command Default** This command is disabled.

**Command Modes** Global configuration (config)

Command History	Release	Modification
	15.0(1)M	This command was introduced in a release earlier than Cisco IOS Release 15.0(1)M.

### Examples

The following example shows how to configure a router to send the IP address in an RTSP setup message:

```
Router# configure terminal
Router(config)# rtsp client rtpsetup enable
```

Related Commands	Command	Description
	<b>rtsp client session history duration</b>	Specifies how long to keep RTSP client history records in memory.
	<b>rtsp client timeout connect</b>	Sets the number of seconds allowed for the router to establish a TCP connection to an RTSP server.

## 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*

<b>Syntax Description</b>	<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

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	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
```

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>call application voice load</b>	Allows reload of an application that was loaded via the MGCP scripting package.
	<b>rtsp client session history duration</b>	Specifies the how long the RTSP is kept during the session.
	<b>show call application voice</b>	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**

<b>Syntax Description</b>	<i>seconds</i>	How long, in seconds, the router waits to connect to the server before timing out. Range is 1 to 20.
---------------------------	----------------	--

**Command Default** 3 seconds

**Command Modes** Global configuration

<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.2(11)T	This command was introduced.

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

**Examples** The following example sets the connection timeout to 10 seconds:

```
rtsp client timeout connect 10
```

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>rtsp client session history records</b>	Sets the maximum number of records to store in the RTSP client session history.
	<b>rtsp client timeout message</b>	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

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

### Usage Guidelines

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

### Examples

The following example sets the request timeout to 10 seconds:

```
rtsp client timeout message 10
```

### Related Commands

Command	Description
<b>rtsp client session history records</b>	Sets the maximum number of records to store in the RTSP client session history.
<b>rtsp client timeout connect</b>	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*

**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>l match -pattern</i>	Stream editor (SED) expression used to match incoming call information. The slash "/" is a delimiter in the pattern.
<i>l replacement -rule</i>	SED expression used to replace match-pattern in the call information. The slash "/" is a delimiter in the pattern.
<i>l domain -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

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

**Table 1: 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.
/^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 called a 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
<b>show voice enum-match-table</b>	Displays the configuration of a voice ENUM match table.
<b>test enum</b>	Tests the ENUM rule.
<b>voice enum-match-table</b>	Initiates the definition of a voice ENUM match table.

## rule (SIP Profile Configuration)

To tag rules in SIP profile configurations, use **rule** command in voice class sip-profiles configuration mode. To remove a rule from a SIP profile configuration, use **no** form this command.

```
rule before tag request method {sdp-header | sip-header} header-name {add | copy | modify | remove}
string
rule before tag response method {sdp-header | sip-header} header-name {add | copy | modify |
remove} string
no rule tag
```

Syntax Description	
<i>tag</i>	Specifies the rule number. Range is 1 to 1073741823.
<b>before</b>	(Optional) Specifies the position of the new rule in the SIP profile configuration.
<b>request</b>	Modifies a SIP profile to add, copy, modify, or remove a SIP or SDP header value from a SIP request message.
<b>response</b>	Modifies a SIP profile to add, copy, modify, or remove a SIP or SDP header value from a SIP response message.
<i>method</i>	Type of message to be added, modified, or removed. It can be one of the following values: <ul style="list-style-type: none"> <li>• <b>ack</b> --SIP acknowledgment message.</li> <li>• <b>any</b> --Any SIP message.</li> <li>• <b>bye</b> --SIP BYE message.</li> <li>• <b>cancel</b> --SIP CANCEL message.</li> <li>• <b>comet</b> --SIP COMET message.</li> <li>• <b>info</b> --SIP INFO message.</li> <li>• <b>invite</b> --The first SIP INVITE message.</li> <li>• <b>notify</b> --SIP NOTIFY message.</li> <li>• <b>options</b> --SIP OPTIONS message.</li> <li>• <b>prack</b> --SIP PRACK message.</li> <li>• <b>publish</b> --SIP PUBLISH message.</li> <li>• <b>refer</b> --SIP REFER message.</li> <li>• <b>register</b> --SIP REGISTER message.</li> <li>• <b>reinvite</b> --SIP REINVITE message.</li> <li>• <b>subscribe</b> --SIP SUBSCRIBE message.</li> <li>• <b>update</b> --SIP UPDATE message.</li> </ul>

<b>sdp-header</b>	Specifies an SDP header.
<b>sip-header</b>	Specifies a SIP header.
<i>header-name</i>	SDP or SIP header name.
<b>add</b>	Adds a header.
<b>copy</b>	Copies a header.
<b>modify</b>	Modifies a header.
<b>remove</b>	Removes a header.
<i>string</i>	String to be added, copied, modified, or removed as a header.
<b>Note</b>	If you use the <b>copy</b> keyword, you must provide a matching pattern followed by the variable name for the <i>string</i> argument.

**Command Default** SIP profile configurations are in non-rule format.

**Command Modes** Voice class configuration (config-class)

Command History	Release	Modification
	15.5(2)T, Cisco IOS XE Release 3.15S	This command was introduced.

**Usage Guidelines** This command tags the rules in a SIP profile configuration. The **before** keyword is used to introduce a new command at any position in the existing set of rules in a SIP profile configuration.

### Example

Example for tagging a SIP profile rule

```
Device(config)# voice class sip-profiles 10
Device(config-class)# rule 1 request invite sip-header contact copy "(.*)" u01
```

Example for inserting a rule in an existing SIP profile

```
Device(config)# voice class sip-profiles 10
Device(config-class)# rule before 1 request invite sip-header contact copy "(.*)" u01
```

### Related Commands

Command	Description
<b>request</b>	Modifies a SIP profile to add, copy, modify, or remove a SIP or SDP header value from a SIP request message.
<b>response</b>	Modifies a SIP profile to add, copy, modify, or remove a SIP or SDP header value from a SIP response message.



## 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.
<b>type</b> match -type replace-type	<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"> <li>• <b>abbreviated</b> --Abbreviated representation of the complete number as supported by this network.</li> <li>• <b>any</b> --Any type of called number.</li> <li>• <b>international</b> --Number called to reach a subscriber in another country.</li> <li>• <b>national</b> --Number called to reach a subscriber in the same country, but outside the local network.</li> <li>• <b>network</b> --Administrative or service number specific to the serving network.</li> <li>• <b>reserved</b> --Reserved for extension.<b>subscriber</b>--Number called to reach a subscriber in the same local network.</li> <li>• <b>unknown</b> --Number of a type that is unknown by the network.</li> </ul> <p>Valid values for the <i>replace-type</i> argument are as follows:</p> <ul style="list-style-type: none"> <li>• <b>abbreviated</b> --Abbreviated representation of the complete number as supported by this network.</li> <li>• <b>international</b> --Number called to reach a subscriber in another country.</li> <li>• <b>national</b> --Number called to reach a subscriber in the same country, but outside the local network.</li> </ul>

<b>type</b> <i>match -type</i> <i>replace-type</i> (continued)	<ul style="list-style-type: none"> <li>• <b>network</b> --Administrative or service number specific to the serving network.</li> <li>• <b>reserved</b> --Reserved for extension.</li> <li>• <b>subscriber</b> --Number called to reach a subscriber in the same local network.</li> <li>• <b>unknown</b> --Number of a type that is unknown by the network.</li> </ul>
<b>plan</b> <i>match -type</i> <i>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"> <li>• <b>any</b> --Any type of dialed number.</li> <li>• <b>data</b></li> <li>• <b>ermes</b></li> <li>• <b>isdn</b></li> <li>• <b>national</b> --Number called to reach a subscriber in the same country, but outside the local network.</li> <li>• <b>private</b></li> <li>• <b>reserved</b> --Reserved for extension.</li> <li>• <b>telex</b></li> <li>• <b>unknown</b> --Number of a type that is unknown by the network.</li> </ul> <p>Valid values for the <i>replace-type</i> argument are as follows:</p> <ul style="list-style-type: none"> <li>• <b>data</b></li> <li>• <b>ermes</b></li> <li>• <b>isdn</b></li> <li>• <b>national</b> --Number called to reach a subscriber in the same country, but outside the local network.</li> <li>• <b>private</b></li> <li>• <b>reserved</b> --Reserved for extension.</li> <li>• <b>telex</b></li> <li>• <b>unknown</b> --Number of a type that is unknown by the network.</li> </ul>
<b>reject</b>	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.
15.1(4)M	This command was introduced with an increase in the maximum value of the precedence variable from 15 to 100.

**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.

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

**Table 2: 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
<b>show voice translation-rule</b>	Displays the parameters of a translation rule.
<b>voice translation-rule</b>	Initiates the voice translation-rule definition.