SIP: Busy Out Support

First Published: February 27, 2006
Last Updated: February 27, 2006

The SIP: Busy Out Support feature introduces, at the SIP level, a generic keepalive mechanism that allows the SIP gateway to monitor the status of the SIP servers and provide the option of busying-out the associated voice ports upon total keepalive failure.

Note Generic means that the keepalive mechanism work with any SIP server, not just Cisco equipment.

Finding Feature Information in This Module

Your Cisco IOS software release may not support all of the features documented in this module. To reach links to specific feature documentation in this module and to see a list of the releases in which each feature is supported, use the "Feature Information for SIP: Busy Out Support" section on page 20.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

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Prerequisites for SIP: Busy Out Support

For information about configuring voice functionality, see the *Cisco IOS Voice Configuration Library*.

- Establish a working IP network.
- Configure VoIP.
- Ensure that the gateway has voice functionality configured for SIP.
- Be familiar with the Cisco IOS SIP Configuration Guide chapters entitled, "Overview of SIP" and "Basic SIP Configuration."

Information About SIP: Busy Out Support

In order to use the SIP Busy Out feature, you should understand the following concepts:

- SIP: Busy Out Support Functionality, page 2
- Benefit of the SIP: Busy Out Support Feature, page 3

SIP: Busy Out Support Functionality

The SIP: Busy Out Support feature allows you to configure a SIP gateway to monitor the status of the SIP server by using a keepalive mechanism. When a lack of keepalive responses from a SIP server indicate a failure, the configured voice ports present a seized or busied-out condition to the attached private branch exchange (PBX) or other customer premises equipment (CPE). The SIP: Busy Out Support feature causes calls to fallback to another means of connection more quickly in the event of a failure than if the feature were not enabled. For example, if the connection between the client and a SIP server goes down and a call is made, an INVITE message request is not sent to the downed SIP server, instead, the system sees that the link is down and looks for alternate routes. The PBX or CPE then attempts to select an alternate route. When a voice port is busied out, the SIP gateway resumes the keepalive mechanism and unbusys the associated voice ports upon receipt of a message response.

The SIP: Busy Out Support feature differs from existing connection admission control (CAC) mechanisms because it works at the SIP level and does not require any proprietary capabilities on the remote SIP server. The SIP: Busy Out Support feature works over channel associated signaling (CAS), Primary Rate Interface (PRI), and Foreign Exchange Station (FXS). In addition, the SIP: Busy Out Support feature adds a new command at the dial-peer level to prevent registration of selected plain old telephone service (POTS) dial-peers.

The SIP: Busy Out Support feature involves the following events that help explain how the feature works:

- Options Message Requests Sent to an Active SIP Server or Servers, page 2
- Options Message Requests Sent to a Server Not Responding, page 3

Options Message Requests Sent to an Active SIP Server or Servers

The **keepalive target** command must be enabled within the SIP UA configuration mode, so that the SIP gateway can send Options message requests as a keepalive mechanism to active SIP servers. The SIP gateway sends Options message requests to the specified SIP servers at the rate specified by the **timers keepalive**

active command active timer. As long as the SIP gateway receives responses to these Options message requests and the command is not disabled, the SIP gateway continues sending Options message requests at this rate.

However, if the configured SIP servers fail to respond to an Options message request by the time the configurable **retry keepalive** count has been exhausted, the configured voice port or ports, are busied out and the SIP gateway begins sending Options message requests at the rate specified under the **timers keepalive down** command.

Busy Voice Ports

Voice ports are made busy in a manner similar to that used with gatekeeper monitoring in H.323 (Enhanced Features for Local and Advanced Voice Busyout). When the keepalive mechanism fails and the SIP servers are in the down state, the gateway. cycles through the voice ports and shuts down voice ports that are configured to monitor SIP keepalives.

The callback register invokes the voice port busyout action and sets the ABCD bit to reflect busy in the case of channel associated signaling (CAS) and generates busy D signal in common channel signaling (CCS). If no voice ports have busyout monitor keepalive configured, the Options message request are sent as long as the **keepalive target** command is configured, but no voice ports are busied or unbusied.

Options Message Requests Sent to a Server Not Responding

When the SIP servers fail to respond in the down state, the Options message requests continue to be sent, at the rate specified by the **timers keepalive down** command. At the time of failure, the voice ports are cycled through, and any voice ports with **busyout monitor** command keepalive configured are busied out. When the SIP gateway receives a sufficient configurable number of Options message responses, the Options message requests resumed at the rate specified by the **timers keepalive active** command. When the servers resume the active state, the configured voice ports are unbusied. The SIP gateway does not unbusy the ports until the number of sequential responses specified via the **keepalive trigger** command are received. This prevents flapping or repeatedly busying and unbusying the ports.

Unbusy Voice Ports

The voice ports are removed from the unbusy state in a manner similar to that used in gatekeeper monitoring in H.323 (Enhanced Features for Local and Advanced Voice Busyout). When the keepalive mechanism becomes active (from the down state), the voice ports that were busied because of keepalive failurea are unbusied. Ports with a matching busyout type are unbusied. This callback handles setting the ABCD bit and D channel signal to reflect not-busy in the case of CAS and CCS, respectively.

Benefit of the SIP: Busy Out Support Feature

The SIP: Busy Out Support feature makes fallback seamless to the user.

For example; you have two dial peers configured for calling 555-1212. One of the dial peers routes the outbound call to 555-1212 through SIP server A, and the other routes it through SIP server B (both routes end up in the same place). If SIP server A has a higher preference than SIP server B, it is always tried first when someone dials 555-1212, and it fails, the SIP gateway tries SIP server B via the other dial peer. However, if the SIP: Busy Out Support feature is implemented and it detects a failure in the connection with SIP server A, it marks that connection as unavailable (before anyone dials the number) so that it is skipped when the number is dialed, going straight to the second dial peer (SIP server B), thus saving the time of waiting for the call through SIP server A to fail. The end result is that someone placing an outbound call does not have to wait as long for the call to complete if there are multiple routes available for that particular call, and one route is down.

How to Configure the SIP: Busy Out Support Feature

This section contains the following procedure:

- Configuring the SIP: Busy Out Support Feature, page 4 (required)

Configuring the SIP: Busy Out Support Feature

Configuring the SIP: Busy Out Support feature involves the following general steps, which are provided in summary form and in detailed form in this section:

- Select voice ports to be made busy in cases of a keepalive failure.
- Set where the keepalives are to be sent; to either one or two SIP servers depending on how many are configured
- Set the time in seconds between sending Options message requests when the SIP servers are active or down state.
- Set the retry count for the keepalive retransmissions.
- Set the number of Options message requests that must be consecutively received from the SIP servers in order to unbusy the voice ports when in the SIP servers are in the down state.
- Verify the busyout ports

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port*
4. **busyout monitor keepalive**
5. **sip-ua**
6. **keepalive target** {**ipv4:***address*[**:***port*] | **dns:***hostname*} [**tcp** [**tls**] | **udp**] [**secondary**]
7. **timers keepalive active** | **down** *seconds*
8. **retry keepalive** *count*
9. **keepalive trigger** *count*
10. **end**
11. **show voice busyout**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *port* **Example:** `Router(config)# voice-port 2/2`	Enters voice-port configuration mode. To find the port argument for your router, see the Cisco IOS Voice Command Reference, Release 12.4T.
Step 4	**busyout monitor keepalive** **Example:** `Router(config-voiceport)# busyout monitor keepalive`	Selects a voice port to be busied-out in case of a keepalive failure.
Step 5	**sip-ua** **Example:** `Router(config-dial-peer)# sip-ua`	Enters SIP user-agent configuration mode.
Step 6	**keepalive target {ipv4:***address*[**:***port*] | **dns:***hostname*} [**tcp** [**tls**] | **udp**] [**secondary**] **Example:** `Router(config-sip-ua)# keepalive target ipv4:172.16.0.2 tcp secondary`	Sets the keepalive target to either primary, secondary, or both depending on how many are configured. • **ipv4:***address*—IP address (in IP version 4 format) of the primary or secondary SIP server to monitor • **:***port*—(Optional) SIP port number. Default SIP port number is 5060. • **dns:***hostname*—DNS hostname of the primary or secondary SIP server to monitor • **udp**—(Optional) Sends keepalive over UDP • **tcp**—(Optional) Sends keepalive over TCP • **tls**—(Optional) Sends keepalive over TLS • **secondary**—(Optional) Sets the secondary server address
Step 7	**timers keepalive active** *seconds* **Example:** `Router(config-sip-ua)# timers keepalive active 20`	(Optional) Sets the keepalive active timer value in seconds. • *seconds*—Keepalive active timer value in seconds. Range: 10 to 600. Default: 120.

	Command or Action	Purpose
Step 8	**retry keepalive** *count* **Example:** `Router(config-sip-ua)# retry keepalive 5`	(Optional) Sets the retry count for keepalive retransmissions. • *count*— Retry count for keepalive retransmissions. Range: 1 to 10. Default: 6.
Step 9	**keepalive trigger** *count* **Example:** `Router(config-sip-ua)# keepalive trigger 5`	(Optional) Sets the keepalive trigger count. • *count*— Keepalive trigger count. Range: 1 to 10. Default: 3.
Step 10	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.
Step 11	**show voice busyout** **Example:** `Router# show voice busyout`	(Optional) Displays information about the voice-busyout states.

Examples

The following sample output of the **show voice busyout** command shows which voice ports are in busy out state:

```
Router# show voice busyout

If following network interfaces are down, voice port will be put into busyout state
ATM0
Serial0
Voice port busyout will be triggered by the following network states

1/0/0   busyout monitor keepalive

The following voice ports are in busyout state

1/1      is forced into busyout state
1/2      is in busyout state caused by network interfaces
1/3      is in busyout state caused by ATM0
1/4      is in busyout state caused by network interfaces
1/5      is in busyout state caused by Serial0
```

Configuration Examples for SIP: Busy Out Support

The following shows examples of the SIP: Busy Out Support feature when enabled.

Examples of keepalive information shown are as follows:

- Busyout monitor command
- Keepalive target command with the primary server using an IP version 4 address
- Keepalive target command with the secondary server using a DNS hostname
- Keepalive trigger command

- Keepalive retry command

```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname chef
!
boot-start-marker
boot-end-marker
!
no logging buffered
!
no aaa new-model
!
resource policy
!
clock timezone GMT 5
clock summer-time Gmt recurring
!
!
--More--
ip cef
ip domain name sip.com
ip name-server 172.18.192.48
!
!
voice call debug full-guid
!
voice service voip
 sip
 registrar server expires max 600 min 60
!
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
!
!
--More--
!
application
 service voipapp tftp://172.18.207.15/gw-tcl-scripts/mirinda/mirinda_regress_server.tcl
!
!
username jsmith password 0 jsmith
username cisco password 0 lab
!
!
interface FastEthernet0/0
 ip address 172.18.193.120 255.255.255.0
 duplex auto
 speed auto
 ip rsvp bandwidth 7500 7500
!
--More--
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
```

```
ip http server
!
ip route 0.0.0.0 0.0.0.0 172.18.193.1
!
ip rtcp report interval 200
!
!
control-plane
!
call treatment on
!
!
call filter match-list 1 voice
 incoming dialpeer 125
 outgoing signaling remote ipv4 172.16.0.4
 --More--
voice-port 1/0/0
! Busyout monitor command with keepalive option selected
 busyout monitor keepalive
!
voice-port 1/0/1
!
voice-port 1/1/0
!
voice-port 1/1/1
!
voice-port 2/0/0
!
voice-port 2/0/1
!
!
dial-peer cor custom
!
!
dial-peer voice 1 pots
 destination-pattern 36602
 --More--
 port 2/0/0
! Unregistered dial peer command required for the keepalive mechanism to function
 no sip-register
!
dial-peer voice 6 voip
 destination-pattern 36601
 modem passthrough nse codec g711ulaw
 session protocol sipv2
 session target ipv4:172.18.193.98:5060
 incoming called-number 36602
 dtmf-relay rtp-nte
 codec g711ulaw

!
dial-peer voice 5 voip
 destination-pattern 5550199
 session protocol sipv2
 session target ipv4:172.18.197.182
 codec g711ulaw
!
dial-peer voice 4 voip
 destination-pattern 9002
 session protocol sipv2
 session target ipv4:172.18.193.87
!
 --More--
 dial-peer voice 9001 voip
```

```
 destination-pattern 9001
 session protocol sipv2
 session target ipv4:172.18.195.49
!
dial-peer voice 31 voip
 destination-pattern 3100801
 signaling forward none
 session protocol sipv2
 session target ipv4:64.102.17.208
 codec g711ulaw
!
dial-peer voice 123 voip
 shutdown
 destination-pattern 3100802
 session protocol sipv2
 session target ipv4:172.18.193.99
 codec g711ulaw
!
dial-peer voice 8 voip
 description stan
 shutdown
 destination-pattern 9876
 --More--
 session protocol sipv2
 session target dns:stan-a.sip.com
!
dial-peer voice 333 voip
 destination-pattern 111
 session protocol sipv2
 session target ipv4:172.18.201.177
 codec g711ulaw
!
dial-peer voice 2 pots
 preference 2
 destination-pattern 362
 port 2/0/1
!
dial-peer voice 362 voip
 preference 1
 destination-pattern 362
 session protocol sipv2
 session target ipv4:172.18.193.120
 incoming called-number 362
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco
!
gateway
 --More--
 timer receive-rtp 1200
!

sip-ua
! Keepalive target command with a primary server keepalive mechanism using an IP version 4
address
 keepalive target ipv4:172.18.198.184
! Keepalive target command with a secondary server keepalive mechanism using a DNS
hostname
 keepalive target dns:wendy.sip.com tcp secondary
! Keepalive trigger command set to five
 keepalive trigger 5
 retry invite 2
 retry response 10
 retry bye 2
 retry prack 8
 retry notify 2
```

```
 retry subscribe 8
! Retry keepalive command set to nine
 retry keepalive 9
 sip-server dns:sip.com
!
!
telephony-service
 mwi relay
 max-conferences 8 gain -6
 transfer-system full-consult
!
!
line con 0
--More--
exec-timeout 0 0
line aux 0
line vty 0 4
 login
!
!
end
```

Additional References

The following sections provide references related to the SIP: Busy Out Support feature.

Related Documents

Related Topic	Document Title
SIP	*Cisco IOS SIP Configuration Guide*
	Cisco IOS Voice Configuration Library

Standards

Standard	Title
None	—

MIBs

MIB	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature.	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 3261	*SIP: Session Initiation Protocol*

Technical Assistance

Description	Link
The Cisco Technical Support & Documentation website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/techsupport

Command Reference

This section documents the following new and modified commands:

busyout monitor

To place a voice port into the busyout monitor state, enter the **busyout monitor** command in voice-port configuration mode. To remove the busyout monitor state from the voice port, use the **no** form of this command.

busyout monitor {**serial** *interface-number* | **ethernet** *interface-number* | **keepalive**} [**in-service**]

no busyout monitor {**serial** *interface-number* | **ethernet** *interface-number* | **keepalive**}

Syntax Description

serial	Specifies monitoring of a serial interface. More than one interface can be entered for a voice port.
ethernet	Specifies monitoring of an Ethernet interface. More than one interface can be entered for a voice port.
interface-number	The interface to be monitored for the voice port busyout function.
keepalive	In case of keepalive failures, the selected voice port or ports is busied out.
in-service	(Optional) Configures the voice port to be busied out when any monitored interface comes into service (its state changes to up). If the keyword is not entered, the voice port is busied out when all monitored interfaces go out of service (their state changes to down).

Defaults

The voice port does not monitor any interfaces.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco MC3810.
12.0(5)XE	This command was implemented on the Cisco 7200 series.
12.0(5)XK	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.0(7)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series and integrated into Cisco IOS Release 12.0(7)T.
12.0(7)XK	The ability to monitor an Ethernet port was introduced and the **in-service** keyword was added. The **serial** keyword was first supported on the Cisco 2600 series and Cisco 3600 series.
12.1(1)T	The implementation of this command on the Cisco 7200 series was integrated into Cisco IOS Release 12.1(1)T.
12.1(2)T	The **serial** and **ethernet** keywords were added, the **in-service** keyword was integrated into Cisco IOS Release 12.1(2)T, and the *interface-number* argument was added to the **serial** and **ethernet** keywords.
12.1(3)T	The **interface** keyword was removed.
12.4(6)T	The **keepalive** keyword was added.

Usage Guidelines

When you place a voice port in the busyout monitor state, the voice port monitors the specified interface and enters the busyout state when the interface is down. This down state forces the rerouting of calls.

The **busyout monitor** command monitors only the up or down status of an interface—not end-to-end TCP/IP connectivity.

When an interface is operational, a busied-out voice port returns to its normal state.

This feature can monitor LAN, WAN, and virtual subinterfaces.

A voice port can monitor multiple interfaces at the same time. To configure a voice port to monitor multiple interfaces, reenter the **busyout monitor** command for each additional interface to be monitored.

If you specify more than one monitored interface for a voice port, all the monitored interfaces must be down to trigger busyout on the voice port.

You can combine in-service and out-of-service monitoring on a voice port. The following rule describes the action if monitored interfaces change state. A voice port is busied out if either of the following occurs:

- Any interface monitored for coming into service comes up.
- All interfaces monitored for going out of service go down.

Examples

The following example shows configuration of analog voice port 1/2 to busy out if serial port 0 or 1 comes into service:

```
voice-port 1/2
 busyout monitor serial 0 in-service
 busyout monitor serial 1 in-service
```

The following example shows configuration of digital voice port 1/2/2 on a Cisco 3600 series router to busy out if serial port 0 goes out of service:

```
voice-port 1/2/2
 busyout monitor serial 0
```

The following example shows configuration of the voice port to monitor two serial interfaces and an Ethernet interface. When all these interfaces are down, the voice port is busied out. When at least one interface is operating, the voice port is put back into a normal state.

```
voice-port 3/0:0
 busyout monitor ethernet 0/0
 busyout monitor serial 1/0
 busyout monitor serial 2/0
```

The following example shows configuration of the voice port to be busied out in case of a keepalive failure:

```
voice-port 10
 busyout monitor keepalive
```

Related Commands

Command	Description
busyout forced	Forces a voice port into the busyout state.
busyout monitor probe	Configures a voice port to enter busyout state if an SAA probe signal returned from a remote interface crosses a delay or loss threshold.

busyout seize	Changes the busyout seize procedure for a voice port.
show voice busyout	Displays information about the voice busyout state.
voice-port busyout	Places all voice ports associated with a serial or ATM interface into a busyout state.

keepalive target

To identify session initiation protocol (SIP) servers that will receive keepalive packets from the SIP gateway, use the **keepalive target** command in SIP user agent configuration mode. To disable the **keepalive target** command behavior, use the **no** form of this command.

keepalive target {**ipv4:***address*[**:***port*] | **dns:***hostname*} [**tcp** [**tls**]] | **udp**] [**secondary**]

no keepalive target [**secondary**]

Syntax Description

ipv4:*address*	IP address (in IP version 4 format) of the primary or secondary SIP server to monitor.
:*port*	(Optional) SIP port number. Default SIP port number is 5060.
dns:*hostname*	DNS hostname of the primary or secondary SIP server to monitor.
tcp	(Optional) Sends keepalive packets over TCP.
tls	(Optional) Sends keepalive packets over TLS.
udp	(Optional) Sends keepalive packets over User Datagram Protocol (UDP).
secondary	(Optional) Associates the IP version 4 address or the DNS hostname to a secondary SIP server to monitor.

Command Default

No keepalives are sent by default from SIP gateway to SIP gateway. The SIP port number is 5060 by default.

Command Modes

SIP user agent configuration

Command History

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

Usage Guidelines

The primary or secondary SIP server addresses are in the following forms: dns:example.sip.com or ipv4:172.16.0.10.

Examples

The following example sets the primary SIP server address and defaults to the UDP transport:

```
sip-ua
 keepalive target ipv4:172.16.0.10
```

The following example sets the primary SIP server address and the transport to UDP:

```
sip-ua
 keepalive target ipv4:172.16.0.10 udp
```

The following example sets both the primary and secondary SIP server address and the transport to UDP:

```
sip-ua
```

```
 keepalive target ipv4:172.16.0.10 udp
 keepalive target ipv4:172.16.0.20 udp secondary
```

The following example sets both the primary and secondary SIP server addresses and defaults to the UDP transport:

```
sip-ua
 keepalive target ipv4:172.16.0.10
 keepalive target ipv4:172.16.0.20 secondary
```

The following example sets the primary SIP server address and the transport to TCP:

```
sip-ua
 keepalive target ipv4:172.16.0.10 tcp
```

The following example sets both the primary and secondary SIP server addresses and the transport to TCP:

```
sip-ua
 keepalive target ipv4:172.16.0.10 tcp
 keepalive target ipv4:172.16.0.20 tcp secondary
```

The following example sets the primary SIP server address and the transport to TCP and sets security to TLS mode:

```
sip-ua
 keepalive target ipv4:172.16.0.10 tcp tls
```

The following example sets both the primary and secondary SIP server addresses and the transport to TCP and sets security to the TLS mode:

```
sip-ua
 keepalive target ipv4:172.16.0.10 tcp tls
 keepalive target ipv4:172.16.0.20 tcp tls secondary
```

Related Commands

Command	Description
busyout monitor keepalive	Selects a voice port or ports to be busied out in cases of a keepalive failure.
keepalive trigger	Sets the trigger count 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.
retry keepalive	Sets the retry keepalive count for retransmission.
timers keepalive	Sets the timers keepalive interval between sending Options message requests when the SIP server is active or down.

keepalive trigger

The trigger count represents the number of Options message requests that must consecutively receive responses from the SIP servers when in the down state in order to unbusy the voice ports, use the **keepalive trigger** command in SIP user agent configuration mode. To restore to the default value of 3 seconds, use the **no** form of this command.

keepalive trigger *count*

no keepalive trigger *count*

Syntax Description

count	Keepalive trigger value in the range from 1 to 10. The default value is 3.

Command Default

The default value for the keepalive trigger is 3.

Command Modes

SIP user agent configuration

Command History

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

Usage Guidelines

Sets the count to represent the number of Options message requests that must be consecutively receive responses from the SIP servers in order to unbusy the voice ports when in the down state. The default is 3.

Examples

The following example sets a time interval after 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. The trigger interval is set to 8 in the following example:

```
sip-ua
 keepalive trigger 8
```

Related Commands

Command	Description
busyout monitor keepalive	Selects a voice port or ports to be busied out in cases of a keepalive failure.
keepalive target	Identifies a SIP server that will receive keepalive packets from the SIP gateway.
retry keepalive	Sets the retry keepalive for retransmission.
timers keepalive	Sets the time interval between sending Options message requests when the SIP server is active or down.

retry keepalive

To set the retry count for keepalive retransmission, use the **retry keepalive** command in SIP user agent 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

count	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 user agent configuration

Command History

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

Usage Guidelines

Sets the keepalive retransmissions retry count.

Examples

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

```
sip-ua
 retry keepalive 8
```

Related Commands

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

timers keepalive

To set the keepalive timers interval between sending Options message requests when the session initiation protocol (SIP) servers are in the down state, use the **timers keepalive** command in SIP user agent configuration mode. To restore the keepalive timers to the default value of 120 seconds when active or 30 seconds when down, use the **no** form of this command.

timers keepalive {**active** | **down**} *seconds*

no timers keepalive {**active** | **down**} *seconds*

Syntax Description

active	SIP servers are in the active state.
down	SIP servers are in the down state.
seconds	Time in seconds between keepalive messages when the SIP servers are either active or down, as follows: • If **active** is specified, the range is from 10 to 600 seconds; the default value is 120 seconds. • If **down** is specified, the range is from 1 to 120 seconds; the default value is 30 seconds.

Command Default

The default value for the active state is 120 seconds and the default value for the down state is 30 seconds.

Command Modes

SIP user agent configuration

Command History

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

Usage Guidelines

Use this command to change the keepalive message time interval in seconds between the sending Options message requests when the SIP server or servers are either in the active or down state.

Examples

The following example sets the keepalive message time interval to 20 seconds when the SIP server is in the active state:

```
sip-ua
 timers keepalive active 20
```

The following example sets the keepalive message time interval to 10 seconds when the SIP server is in the down state:

```
sip-ua
 timers keepalive down 10
```

Related Commands

Command	Description
busyout monitor keepalive	Selects a voice port or ports to be busied out in cases of a keepalive failure.
keepalive target	Identifies a SIP server that will receive keepalive packets from the SIP gateway.
keepalive trigger	Sets 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.
retry keepalive	Sets the retry keepalive count for retransmissions.

Feature Information for SIP: Busy Out Support

Table 1 lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Cisco IOS software images are specific to a Cisco IOS software release, a feature set, and a platform. Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Note Table 1 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Table 1 Feature Information for SIP: Busy Out Support

Feature Name	Releases	Feature Information
SIP: Busy Out Support	12.4(6)T	The SIP: Busy Out Support feature introduces, at the SIP level, a generic keepalive mechanism that allows the SIP gateway to monitor the status of the SIP servers and provide the option of busying-out the associated voice ports upon total keepalive failure. The following sections provide information about this feature: • SIP: Busy Out Support Functionality, page 2 • Benefit of the SIP: Busy Out Support Feature, page 3 • Configuring the SIP: Busy Out Support Feature, page 4 The following commands were introduced or modified by this feature: busyout monitor, keepalive target, keepalive trigger, retry keepalive, and timers keepalive