SIP Gateway Enhancements

The SIP Gateway Enhancements feature describes new features for Session Initiation Protocol (SIP) networks.

Feature Specifications for the SIP Gateway Enhancements

Feature History	
Release	**Modification**
12.2(15)ZJ	This feature was introduced.
Supported Platforms	
Cisco 1751, Cisco 1751V, Cisco 1760, Cisco 2610XM, Cisco 2611XM, Cisco 2620XM, Cisco 2621XM, Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3640, Cisco 3640A, Cisco 3660; Cisco 3725, Cisco 3745	

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 Gateway Enhancements

The following are general prerequisites for SIP deployment:

- Ensure that the gateway has voice functionality that is configurable for SIP.
- Establish a working IP network.

  For more information about configuring IP, refer to the following document:

  Cisco IOS IP Configuration Guide, Release 12.2
- Configure VoIP.

  For more information about configuring VoIP, refer to the following document:

  Cisco IOS Voice, Video, and Fax Configuration Guide, Release 12.2
- The SIP gateway must support 300 or 302 Redirect messages.

Restrictions for SIP Gateway Enhancements

- To support Skinny Client Control Protocol (SCCP) IP phones, originating and terminating SIP gateways can use NOTIFY-based out-of-band dual-tone multifrequency (DTMF) relay. NOTIFY-based out-of-band DTMF relay is a Cisco proprietary function.
- SIP gateways do not support authentication, and therefore cannot respond to authentication requests for Register messages.

Information About SIP Gateway Enhancements

To configure the SIP Gateway Enhancements feature, you must understand the following concepts:

- Call Redirect Enhancements to Support IP-to-IP Calls through the IOS Voice Gateway, page 2
- How to Configure SIP Gateway Enhancements, page 6
- NOTIFY-Based Out-of-Band DTMF Relay, page 3
- SIP Register Support, page 5

Call Redirect Enhancements to Support IP-to-IP Calls through the IOS Voice Gateway

The Cisco IOS Voice Gateway has been enhanced to use call redirection if an incoming VoIP call matches an outbound VoIP dial peer. The gateway sends a 300 or 302 Redirect message to the call originator allowing the originator to reestablish the call.

Two new commands allow you to enable the redirect functionality, globally or on a specific inbound dial peer. For command-level information, see the appropriate command page.

- redirect ip2ip (dial-peer), page 27
- redirect ip2ip (voice service), page 28

Sending 300 Multiple Choice Messages

Prior to Cisco IOS Release 12.2(15)ZJ, when a call was redirected, the SIP gateway would send a “302 Moved Temporarily” message. The first longest match route on a gateway (dial-peer destination pattern) was used in the Contact header of the 302 message. With release 12.2(15)ZJ, if multiple routes to a destination exist for a redirected number (multiple dial peers are matched), the SIP gateway sends a “300 Multiple Choice” message and the multiple routes in the Contact header are listed.

A new command has been added to give users the flexibility to choose the order in which the routes can appear in the Contact header.

- redirect contact order, page 26

NOTIFY-Based Out-of-Band DTMF Relay

SCCP IP phones do not support in-band DTMF digits; they are capable of sending only out-of-band DTMF digits. To support SCCP devices, originating and terminating SIP gateways can use Cisco proprietary NOTIFY-based out-of-band DTMF relay. In addition, NOTIFY-based out-of-band DTMF relay can also be used by analog phones attached to analog voice ports (FXS) on the router.

NOTIFY-based out-of-band DTMF relay sends message bidirectionally between the originating and terminating gateways for a DTMF event during a call. If multiple DTMF relay mechanisms are enabled on a SIP dial peer and are negotiated successfully, NOTIFY-based out-of-band DTMF relay takes precedence.

The originating gateway sends an Invite message with SIP Call-Info header to indicate the use of NOTIFY-based out-of-band DTMF relay. The terminating gateway acknowledges the message with an 18x/200 Response message, also using the Call-Info header. The Call-Info header for NOTIFY-based out-of-band relay appears as follows:

```
Call-Info: <sip: address>; method="NOTIFY;Event=telephone-event;Duration=msec"
```

Note Duration is the interval between NOTIFY messages sent for a single digit and is set through the **notify telephone-event** command.

After the NOTIFY-based out-of-band DTMF relay mechanism is negotiated by the SIP Invite and 18x/200 Response messages, whenever a DTMF event occurs the gateway sends a SIP NOTIFY message for that event. In response, the gateway expects to receive a 200 OK message.

The NOTIFY-based out-of-band DTMF relay mechanism is similar to the DTMF message format described in RFC 2833. It consists of 4 bytes in a binary encoded format.

Figure 1 Message Format of NOTIFY-based out-of-band DTMF relay

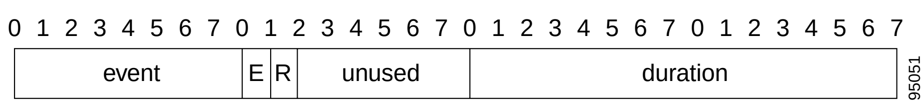

Table 1

Field	Description
event	The DTMF event that is between 0-9, A, B, C, D, #, * and flash.
E	E signifies the end bit. If E is set to a value of one, the NOTIFY message contains the end of the DTMF event. Thus, the duration parameter in this final NOTIFY message measures the complete duration of the event.
R	Reserved.
unused	In RFC 2833, unused corresponds to the volume field, but is not used in NOTIFY-based out-of-band DTMF relay.
duration	Duration of this DTMF event in milliseconds.

Sending NOTIFY messages

As soon as the DTMF event is recognized, the gateway sends out an initial NOTIFY message for this event with the duration negotiated in the Invite's Call-Info header. For the initial NOTIFY message the end bit is set to zero. Afterward, one of the following can happen:

- If the entire duration of the DTMF event is less than the negotiated duration, the originating gateway sends an end NOTIFY message for this event with the duration field containing the exact duration of the event and the end bit set to one.
- If the duration of the DTMF event is greater than the negotiated duration, then the originating gateway sends another NOTIFY message for this event after the initial timer runs out. The updated NOTIFY message has a duration of twice the negotiated duration. The end bit is set to zero since the event is not yet over. If the event lasts beyond the duration specified in the first updated NOTIFY message, then another updated NOTIFY message is sent with three times the negotiated duration.
- This continues until the DTMF event ends, and an end NOTIFY message is sent. If the event lasts for exactly the negotiated duration, then either of the above two cases can happen, based on whether the end of DTMF event occurred earlier or the timer ran out earlier, respectively.

For example, if the negotiated duration is 600ms, then as soon as a DTMF event occurs, the initial NOTIFY message is sent with duration as 600ms. Then a timer is started for this duration.

- If the DTMF event last only for 300ms, the timer is stopped and an end NOTIFY message is sent with the duration as 300ms.
- If the DTMF event lasts for more than 600ms (1000ms), when the timer expires an updated NOTIFY message is sent with the duration as 1200ms and the timer is restarted. Finally when the DTMF event ends, an end NOTIFY message is sent with the duration set to 1000ms.

Every DTMF event corresponds to at least two NOTIFYs: an initial NOTIFY message and an end NOTIFY. There might be some update NOTIFYs also involved, if the total duration of the event is greater than the negotiated max-duration interval. Since DTMF events generally last for less than 1000ms, setting the duration using **notify telephone-event** command to more than 1000ms reduces the total number of NOTIFY messages sent. The default value of **notify telephone-event** command is 2000ms.

Receiving NOTIFY messages

Once a NOTIFY message is received by the terminating gateway, the DTMF tone plays and a timer is set for the value in the duration field. Afterward, one of the following can happen:

- If an end NOTIFY message for a DTMF event is received, the tone is stopped.
- If an update is received, the timer is updated according to the duration field.
- If an update or end NOTIFY message is not received before the timer expires, the tone is stopped and all subsequent NOTIFY messages for the same DTMF-event or DTMF digit are ignored until an end NOTIFY message is received.
- If a NOTIFY message for a different DTMF event is received before an end NOTIFY message for the current DTMF event is received (which is an unlikely case), the current tone is stopped and the new tone is played. This is an unlikely case because for every DTMF event there needs to be a end NOTIFY message, and unless this is successfully sent and a 200 OK received, the gateway cannot send other NOTIFY messages.

Note In-band tones are not passed while NOTIFY-based out-of-band DTMF relay is used as the DTMF relay method.

Two commands allow you to enable or disable NOTIFY-based out-of-band DTMF-relay on a dial peer. The functionality is advertised to other end using Invite messages if it is enabled by the commands, and must be configured on both the originating and terminating SIP gateways. A third command allows you to verify DTMF relay status. For command-level information, see the appropriate command page.

- dtmf-relay (Voice over IP), page 21
- notify telephone-event, page 24
- show sip-ua status, page 40

SIP Register Support

With H.323, Cisco IOS gateways can register E.164 numbers of a POTS dial peer with a gatekeeper, which informs the gatekeeper of a user's contact information. SIP gateways now allow the same functionality, but with the registration taking place with a SIP proxy or registrar. SIP gateways allow registration of E.164 numbers to a SIP proxy or registrar on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and local SCCP phones.

When registering dial peers with an external registrar, you can also register with a secondary SIP proxy or registrar to provide redundancy. The secondary registration can be used if the primary registrar fails.

Note There are no commands that allow registration between the H.323 and SIP protocols.

Several commands have been added to give the user control over enabling, disabling, and monitoring SIP Register messages. For command-level information, see the appropriate command page.

- registrar, page 29
- retry register, page 31
- show sip-ua register status, page 32
- show sip-ua statistics, page 33

- show sip-ua statistics, page 33
- timers register, page 45

How to Configure SIP Gateway Enhancements

This section contains the following procedures:

- Configuring Call Redirect Enhancements to Support Calls, page 6 (required)
- Configuring Sending 300 Multiple Choice Support, page 8 (required)
- Configuring NOTIFY-Based Out-of-Band DTMF Relay, page 9 (required)
- Configuring SIP Register Support, page 12 (required)

Configuring Call Redirect Enhancements to Support Calls

This feature can be enabled globally or on a dial-peer basis.

- Configuring Call Redirect Enhancements to Support Calls Globally, page 6
- Configuring Call Redirect Enhancements to Support Calls on a Specific VoIP Dial Peer, page 7

Configuring Call Redirect Enhancements to Support Calls Globally

To enable global IP-to-IP call redirection for all VoIP dial peers, use voice service configuration mode. The default application on SIP SRST supports IP-to-IP redirection.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **redirect ip2ip**
5. **exit**

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.

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 4	**redirect ip2ip** **Example:** `Router(config-voi-srv)# redirect ip2ip`	Redirects SIP phone calls to SIP phone calls globally on a gateway using the Cisco IOS Voice Gateway.
Step 5	**exit** **Example:** `Router(config-voi-srv)# exit`	Exits voice service configuration mode.

Configuring Call Redirect Enhancements to Support Calls on a Specific VoIP Dial Peer

To specify IP-to-IP call redirection for a specific VoIP dial peer, configure on an inbound dial peer in dial peer configuration mode. The default application on SIP SRST supports IP-to-IP redirection.

Note When IP-to-IP redirection is configured in dial-peer configuration mode, the configuration on the specific inbound dial peer takes precedence over the global configuration entered under voice service configuration.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **application** *application-name*
5. **redirect ip2ip**
6. **exit**

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.

	Command or Action	Purpose
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 25 voip`	Enters dial-peer configuration mode. The *tag* value is a tag that uniquely identifies the dial peer. (This number has local significance only.) The keyword that can be used for configuring redirect is the following: • **voip**—Indicates that this is a VoIP peer using voice encapsulation on the POTS network.
Step 4	**application** *application-name* **Example:** `Router(config-dial-peer)# application session`	Enables a specific application on a dial peer. • For SIP, the default TCL application (from the Cisco IOS image) is **session** and can be applied to both VoIP and POTS dial peers. • The application must support IP-to-IP redirection.
Step 5	**redirect ip2ip** **Example:** `Router(config-dial-peer)# redirect ip2ip`	Redirects SIP phone calls to SIP phone calls on a specific VoIP dial peer using the Cisco IOS Voice Gateway.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.

Configuring Sending 300 Multiple Choice Support

If multiple routes to a destination exist for a redirected number (multiple dial peers are matched), the SIP gateway sends a "300 Multiple Choice" message and the multiple routes in the Contact header are listed. This configuration allows users to choose the order in which the routes appear in the Contact header.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **redirect contact order** [**best-match** | **longest-match**]
6. **exit**

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 service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 4	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 5	**redirect contact order** [**best-match** | **longest-match**] **Example:** `Router(conf-serv-sip)# redirect contact order best-match`	Sets the order of contacts in the 300 Multiple Choice message. The keywords are defined as follows: • **best-match**—Uses the current system configuration to set the order of contacts. • **longest-match**—Sets the contact order by using the destination pattern longest match first, and then the second longest match, the third, and so on. This is the default.
Step 6	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits SIP configuration mode.

Configuring NOTIFY-Based Out-of-Band DTMF Relay

Cisco proprietary NOTIFY-based out-of-band DTMF relay adds support for devices that do not support in-band DTMF. This configuration must be done on both originating and terminating gateways. With this configuration, DTMF tones are forwarded by using SIP NOTIFY messages in SIP Invites or 18x or 200 Response messages.

Restrictions

This configuration can only be done on a SIP VoIP dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **dtmf-relay sip-notify**
5. **exit**
6. **sip-ua**
7. **notify telephone-event max-duration** *time*

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	**`dial-peer voice`** *`tag`* **`voip`** **Example:** `Router(config)# dial-peer voice 123 voip`	Enters dial-peer configuration mode. The *tag* value is a tag that uniquely identifies the SIP dial peer. (This number has local significance only.) The keyword that can be used for configuring redirect on a SIP dial peer is the following: • **voip**—Indicates that this is a VoIP peer using voice encapsulation on the POTS network.
Step 4	**`dtmf-relay sip-notify`** **Example:** `Router(config-dial-peer)# dtmf-relay sip-notify`	Forwards DTMF tones using SIP NOTIFY messages.
Step 5	**`exit`** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.

	Command or Action	Purpose
Step 6	**sip-ua** **Example:** `Router(config)# sip-ua`	Enables SIP user-agent configuration mode.
Step 7	**notify telephone-event max-duration** *time* **Example:** `Router(config-sip-ua)# notify telephone-event max-duration 2000`	Configures the maximum time interval allowed between two consecutive NOTIFY messages for a single DTMF event. The keywords and arguments are defined as follows: • **max-duration** *time*—Time interval between consecutive NOTIFY messages for a single DTMF event, in milliseconds. Range is from 500 to 3000. Default is 2000.

Examples

The following output from the **show running-config** command shows that the **dtmf-relay sip-notify** command is configured in dial peer 123:

```
.
.
.
dial-peer voice 123 voip
 destination-pattern [12]...
 monitor probe icmp-ping
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay sip-notify
.
.
.
```

The following output from the **show sip-ua status** command shows that the time interval between consecutive NOTIFY messages for a telephone event is the default of 2000 ms:

```
Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 6
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED

 SDP application configuration:
 Version line (v=) required
 Owner line (o=) required
 Timespec line (t=) required
 Media supported: audio image
 Network types supported: IN
```

```
Address types supported: IP4
Transport types supported: RTP/AVP udptl
```

Configuring SIP Register Support

SIP gateways allow registration of E.164 numbers to a SIP proxy or registrar server on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and local SCCP phones. By default, SIP gateways do not generate SIP Register messages. The following tasks set up the gateway to register E.164 telephone numbers with an external SIP registrar.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **registrar** {{**dns:** *address* | **ipv4:** *destination-address* } **expires** *seconds* [**tcp**] [**secondary**]}
5. **retry register** *number*
6. **timers register** *time*
7. **end**

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	`sip-ua`
	
	Example:
	`Router(config)# sip-ua`
		Enables SIP user-agent configuration mode.

	Command or Action	Purpose
Step 4	**registrar** {{**dns:** *address* | **ipv4:** *destination-address* } **expires** *seconds* [**tcp**] [**secondary**]} **Example:** `Router(config-sip-ua)# registrar ipv4:10.8.17.40 expires 3600 secondary`	Registers E.164 numbers on behalf of analog telephone voice ports (FXS) and IP phone virtual voice ports (EFXS) with an external SIP proxy or SIP registrar server. • **dns:**:*address*—Domain name server that resolves the name of the dial peer to receive calls. • **ipv4:**—IP address of the dial peer to receive calls. • **Expires** *seconds*—Default registration time, in seconds. • **Secondary**—Specifies registration with a secondary SIP proxy or registrar for redundancy purposes.
Step 5	**retry register** *number* **Example:** `Router(config-sip-ua)# retry register 10`	Sets the total amount of SIP Register messages that the gateway should send. The *number* argument is the number of Register message retries. Range is from 1 to 10. Default is 10.
Step 6	**timers register** *time* **Example:** `Router(config-sip-ua)# timers register 500`	Sets how long the SIP UA waits before sending register requests. The *time* argument is the waiting time, in milliseconds. Range is from 100 to 1000. The default is 500.
Step 7	**end** **Example:** `Router(config-sip-ua)# end`	Exits sip-ua configuration mode.

Examples

The following is sample output from the **show sip-ua timers** command showing the waiting time before a register request is sent; that is, the value that is set with the **timers register** command:

```
Router# show sip-ua timers

SIP UA Timer Values (millisecs)
trying 500, expires 180000, connect 500, disconnect 500
comet 500, prack 500, rel1xx 500, notify 500
refer 500, register 500
```

The following is sample output from the **show sip-ua register status** command showing the status of local E.164 registrations:

```
Router# show sip-ua register status

Line peer expires(sec) registered
4001 20001 596          no
4002 20002 596          no
5100 1     596          no
9998 2     596          no
```

The following is sample output from the **show sip-ua statistics** command showing four registers were sent:

```
Router# show sip-ua statistics
```

```
SIP Response Statistics (Inbound/Outbound)
    Informational:
      Trying 0/0, Ringing 0/0,
      Forwarded 0/0, Queued 0/0,
      SessionProgress 0/0
     Success:
      OkInvite 0/0, OkBye 0/0,
      OkCancel 0/0, OkOptions 0/0,
      OkPrack 0/0, OkPreconditionMet 0/0,
      OkSubscribe 0/0, OkNOTIFY 0/0,
      OkInfo 0/0, 202Accepted 0/0
      OkRegister 12/49
     Redirection (Inbound only except for MovedTemp(Inbound/Outbound)) :
      MultipleChoice 0, MovedPermanently 0,
      MovedTemporarily 0/0, UseProxy 0,
      AlternateService 0
      Client Error:
      BadRequest 0/0, Unauthorized 0/0,
      PaymentRequired 0/0, Forbidden 0/0,
      NotFound 0/0, MethodNotAllowed 0/0,
      NotAcceptable 0/0, ProxyAuthReqd 0/0,
      ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
      ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
      UnsupportedMediaType 0/0, BadExtension 0/0,
      TempNotAvailable 0/0, CallLegNonExistent 0/0,
      LoopDetected 0/0, TooManyHops 0/0,
      AddrIncomplete 0/0, Ambiguous 0/0,
      BusyHere 0/0, RequestCancel 0/0,
      NotAcceptableMedia 0/0, BadEvent 0/0,
      SETooSmall 0/0
     Server Error:
      InternalError 0/0, NotImplemented 0/0,
      BadGateway 0/0, ServiceUnavail 0/0,
      GatewayTimeout 0/0, BadSipVer 0/0,
      PreCondFailure 0/0
     Global Failure:
      BusyEverywhere 0/0, Decline 0/0,
      NotExistAnywhere 0/0, NotAcceptable 0/0
      Miscellaneous counters:
      RedirectRspMappedToClientErr 0

SIP Total Traffic Statistics (Inbound/Outbound)
      Invite 0/0, Ack 0/0, Bye 0/0,
      Cancel 0/0, Options 0/0,
      Prack 0/0, Comet 0/0,
      Subscribe 0/0, NOTIFY 0/0,
      Refer 0/0, Info 0/0
      Register 49/16

Retry Statistics
      Invite 0, Bye 0, Cancel 0, Response 0,
      Prack 0, Comet 0, Reliable1xx 0, NOTIFY 0
      Register 4

SDP application statistics:
Parses: 0, Builds 0
Invalid token order: 0, Invalid param: 0
Not SDP desc: 0, No resource: 0

Last time SIP Statistics were cleared: <never>
```

Configuration Examples for SIP Gateway Enhancements

This section provides the following configuration example.

- SIP Gateway Enhancements Example

Note IP addresses and host names in examples are fictitious.

SIP Gateway Enhancements Example

This section provides a configuration example to match the identified configuration tasks in the previous sections.

```
Current configuration : 3394 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
memory-size iomem 15
ip subnet-zero
!
!
no ip domain lookup
!
ip dhcp pool vespa
 network 192.168.0.0 255.255.255.0
 option 150 ip 192.168.0.1
 default-router 192.168.0.1
!
!
voice call carrier capacity active
!
voice class codec 1
 codec preference 2 g711ulaw
!
!
no voice hpi capture buffer
no voice hpi capture destination
!
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
!
interface Ethernet0/0
 ip address 10.8.17.22 255.255.0.0
 half-duplex
!
interface FastEthernet0/0
 ip address 192.168.0.1 255.255.255.0
 speed auto
 no cdp enable
 h323-gateway voip interface
 h323-gateway voip id vespa2 ipaddr 10.8.15.4 1718
!
router rip
```

```
 network 10.0.0.0
 network 192.168.0.0
!
ip default-gateway 10.8.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.8.0.1
no ip http server
ip pim bidir-enable
!
!
tftp-server flash:SEPDEFAULT.cnf
tftp-server flash:P005B302.bin
call fallback active
!
!
call application global default.new
call rsvp-sync
!
voice-port 1/0
!
voice-port 1/1
!
mgcp profile default
!
!
dial-peer voice 1 pots
 destination-pattern 5100
 port 1/0
!
dial-peer voice 2 pots
 destination-pattern 9998
 port 1/1
!
dial-peer voice 123 voip
 destination-pattern [12]...
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay sip-notify
!
gateway
!
sip-ua
 retry invite 3
 retry register 3
 timers register 150
 registrar dns:myhost3.cisco.com expires 3600
 registrar ipv4:10.8.17.40 expires 3600 secondary
!
!
telephony-service
 max-dn 10
 max-conferences 4
!
ephone-dn 1
number 4001
!
!
ephone-dn 2
number 4002
!
!
line con 0
 exec-timeout 0 0
line aux 0
```

```
line vty 0 4
login
line vty 5 15
 login
!
no scheduler allocate
end
```

Additional References

For additional information regarding the SIP protocol or the SIP Gateway Enhancements feature, refer to the following references.

Related Documents

Related Topic	Document Title
Cisco SIP IP phones	• Cisco IP Phone Documentation for Session Initiation Protocol (SIP)
TCL scripts and programming	• *TCL IVR API Version 2.0 Programming Guide* • "Configuring TCL IVR Applications" chapter in *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
Voice and telephony command reference	*Cisco IOS Voice Command Reference,* Release 12.2 T
Cisco SIP functionality	*Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2 *Session Initiation Protocol (SIP) for VoIP,* Release 12.2(8)T *Session Initiation Protocol Gateway Call Flows,* Release 12.2(4)T

Standards

Standards	Title
No new or modified standards are supported by this feature, and support for existing standards has not been modified by this feature.	—

MIBs

MIBs	MIBs Link
• CISCO-SIP-UA-MIB	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

RFCs	Title
RFC 2543	*SIP: Session Initiation Protocol*
RFC 2833	*RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*

Technical Assistance

Description	Link
Technical Assistance Center (TAC) home page, containing 30,000 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/public/support/tac/home.shtml

Command Reference

This section documents new and modified commands. All other commands used with this feature are documented in the Cisco IOS Release12.3 command reference publications.

New Commands

- **notify telephone-event**
- **redirect contact order**
- **redirect ip2ip (dial-peer)**
- **redirect ip2ip (voice service)**
- **registrar**
- **retry register**
- **show sip-ua register status**
- **timers register**

Modified Commands

- **dtmf-relay (Voice over IP)**
- **show sip-ua statistics**
- **show sip-ua status**
- **show sip-ua timers**

dtmf-relay (Voice over IP)

To specify how an H.323 or Session Initiation Protocol (SIP) gateway relays dual tone multifrequency (DTMF) tones between telephony interfaces and an IP network, use the **dtmf-relay** command in dial-peer configuration mode. To remove all signaling options and send the DTMF tones as part of the audio stream, use the **no** form of this command.

dtmf-relay [**cisco-rtp**] [**h245-alphanumeric**] [**h245-signal**] [**rtp-nte**] [**sip-notify**]

no dtmf-relay [cisco-rtp] [**h245-alphanumeric**] [**h245-signal**] [**rtp-nte**] [**sip-notify**]

Syntax Description

cisco-rtp	(Optional) Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with a Cisco proprietary payload type.
h245-alphanumeric	(Optional) Forwards DTMF tones by using the H.245 “alphanumeric” User Input Indication method. Supports tones from 0 to 9, *, #, and from A to D.
h245-signal	(Optional) Forwards DTMF tones by using the H.245 “signal” User Input Indication method. Supports tones are from 0 to 9, *, #, and from A to D.
rtp-nte	(Optional) Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.
sip-notify	(Optional) Forwards DTMF tones using SIP NOTIFY messages.

Defaults

DTMF tones are disabled and sent inband. That is, they are left in the audio stream.

Command Modes

Dial-peer configuration

Command History

Release	Modification
11.3(2)NA	This command was introduced on the Cisco AS5300.
12.0(2)XH	The **cisco-rtp**, **h245-alphanumeric**, and **h245-signal** keywords were added.
12.0(5)T	This command was integrated into Cisco IOS Release 12.0(5)T.
12.0(7)XK	This command was first supported for VoIP on the Cisco MC3810.
12.1(2)T	Changes made in Cisco IOS Release 12.0(7)XK were integrated into Cisco IOS Release 12.1(2)T.
12.2(8)T	This command was implemented on the Cisco 1751, Cisco 2600 series and Cisco 3600 series, Cisco 3725, and Cisco 3745.
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(4)T	This command does not support the Cisco AS5300, Cisco AS5350, and Cisco AS5400 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.
12.2(15)ZJ	The **sip-notify** keyword was added.

Usage Guidelines

DTMF is the tone generated when you press a button on a touch tone phone. This tone is compressed at one end of a call; when the tone is decompressed at the other end, it can become distorted, depending on the codec used. The DTMF relay feature transports DTMF tones generated after call establishment out-of-band using either a standard H.323 out-of-band method or a proprietary RTP-based mechanism. For SIP calls, the most appropriate methods to transport DTMF tones are RTP-NTE or SIP-NOTIFY.

The SIP-NOTIFY method sends NOTIFY messages bidirectionally between the originating and terminating gateways for a DTMF event during a call. If multiple DTMF relay mechanisms are enabled on a SIP dial peer and are negotiated successfully, the SIP-NOTIFY method takes precedence.

SIP NOTIFY messages are advertised in an Invite message to the remote end only if the **dtmf-relay** command is set.

For SIP, the gateway chooses the format according to the following priority:

1. sip-notify
2. rtp-nte
3. None—DTMF sent in-band

The gateway sends DTMF tones only in the format that you specify if the remote device supports it. If the H.323 remote device supports multiple formats, the gateway chooses the format according to the following priority:

1. cisco-rtp (highest priority)
2. h245-signal
3. h245-alphanumeric
4. rtp-nte
5. None—DTMF sent in-band

The principal advantage of the **dtmf-relay** command is that it sends DTMF tones with greater fidelity than is possible in-band for most low-bandwidth codecs, such as G.729 and G.723. Without the use of DTMF relay, calls established with low-bandwidth codecs may have trouble accessing automated DTMF-based systems, such as voice mail, menu-based Automatic Call Distributor (ACD) systems, and automated banking systems.

Note

- The **cisco-rtp** keyword is a proprietary Cisco implementation and operates only between two Cisco AS5800 access concentrators that are running Cisco IOS Release 12.0(2)XH or between Cisco AS5800 access concentrators or Cisco 2600 series or Cisco 3600 series routers that are running Cisco IOS Release 12.0(2)XH or later releases. Otherwise, the DTMF relay feature does not function, and the gateway sends DTMF tones in-band.
- The **cisco-rtp** keyword of the **dtmf-relay** command is supported on Cisco 7200 series routers.
- The **h245-alphanumeric** and **h245-signal** DTMF settings on a Cisco MC3810 multiservice access concentrator require a high-performance compression module (HCM) and are not supported on a Cisco MC3810 with a non-HCM voice compression module (VCM).
- The **sip-notify** keyword is available only if the VoIP dial peer is configured for SIP.

Examples

The following example configures DTMF relay with the **cisco-rtp** keyword when sending DTMF tones to dial peer 103:

```
dial-peer voice 103 voip
 dtmf-relay cisco-rtp
```

The following example configures DTMF relay with the **cisco-rtp** or **h245-signal** keywords when DTMF tones are sent to dial peer 103:

```
dial-peer voice 103 voip
 dtmf-relay cisco-rtp h245-signal
```

The following example configures the gateway to send DTMF in-band (the default) when DTMF tones are sent to dial peer 103:

```
dial-peer voice 103 voip
 no dtmf-relay
```

The following example configures DTMF relay with the **rtp-nte** keyword when DTMF tones are sent to dial peer 103:

```
dial-peer voice 103 voip
 dtmf-relay rtp-nte
```

The following example configures the gateway to send DTMF tones using SIP NOTIFY messages to dial peer 103:

```
dial-peer voice 103 voip
 session protocol sipv2
 dtmf-relay sip-notify
```

Related Commands

Command	Description
notify telephone-event	Configures the maximum interval between two consecutive NOTIFY messages for a particular telephone-event.

notify telephone-event

To configure the maximum interval between two consecutive NOTIFY messages for a particular telephone-event, use the **notify telephone-event** command in SIP user-agent configuration mode. To reset the interval to the default, use the **no** form of this command.

notify telephone-event max-duration *time*

no notify telephone-event

Syntax Description

max-duration *time*	Time interval between consecutive NOTIFY messages for a single DTMF event, in milliseconds. Range is from 500 to 3000. Default is 2000.

Defaults

2000 milliseconds

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.

Usage Guidelines

The **notify telephone-event** command works with the **dtmf-relay sip-notify** command. The **dtmf-relay sip-notify** command forwards out-of-band DTMF tones by using SIP NOTIFY messages. The **notify telephone-event** command sets the maximum time interval between consecutive NOTIFY messages for a single DTMF event. The maximum time is negotiated between two SIP endpoints and the lowest duration value is the one selected. This duration is negotiated during call establishment as part of negotiating the SIP-NOTIFY DTMF relay.

The originating gateway sends an indication of DTMF relay in an Invite message using the SIP Call-Info header. The terminating gateway acknowledges the message with an 18x/200 Response message, also using the Call-Info header. The set duration appears in the Call-Info header in the following way:

```
Call-Info: <sip: address>; method="Notify;Event=telephone-event;Duration=msec"
```

For example, if the maximum duration of gateway A is set to 1000 ms, and gateway B is set to 700 ms, the resulting negotiated duration would be 700 ms. Both A and B would use the value 700 in all of their NOTIFY messages for DTMF events.

Examples

The following example sets the maximum duration for a DTMF event to 500 ms.

```
Router(config)# sip-ua
Router(config-sip-ua)# notify telephone-event max-duration 500
```

Related Commands

Command	Description
dtmf-relay sip-notify	Forwards DTMF tones using SIP NOTIFY messages.

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 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 to set the order of contacts.
longest-match	(Optional) Sets the contact order by using the destination pattern longest match first, and then the second longest match, the third longest match, and so on. This is the default.

Defaults

longest-match

Command Modes

SIP configuration

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.

Usage Guidelines

This command applies when a “300 Multiple Choice” message is sent by a SIP gateway indicating that the call has been redirected and 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 shows how to set up the **redirect contact order** command to use the current system configuration to set the order of contact:

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

Related Commands

Command	Description
sip	Enters SIP configuration mode from voice service VoIP 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.

Defaults

Disabled

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.

Usage Guidelines

The **redirect ip2ip (dial-peer)** command must be configured on the inbound dial peer of the gateway. This command enables, per dial peer, 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 globally, use **redirect ip2ip (voice service)** command.

Examples

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

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

Related Commands

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

redirect ip2ip (voice service)

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

redirect ip2ip

no redirect ip2ip

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

voice service configuration

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.

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 on a specific inbound dial peer.

Examples

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

```
voice service voip
 redirect ip2ip
```

Related Commands

Command	Description
redirect ip2ip (dial-peer)	Redirects SIP phone calls to SIP phone calls on a specific VoIP dial peer using the Cisco IOS Voice Gateway.

registrar

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

registrar {{**dns:** *address* | **ipv4:** *destination-address*} **expires** *seconds* [**tcp**] [**secondary**]}

no registrar [secondary]

Syntax Description

dns: *address*	DNS address of the SIP Registrar server.
ipv4:*destination-address*	IP address of the SIP Registrar server.
expires *seconds*	Default registration time, in seconds. The range is from 60 to 65535 seconds.
tcp	(Optional) Sets transport layer protocol to TCP. UDP is the default.
secondary	(Optional) Specifies registration with a secondary SIP proxy or registrar to provide redundancy if the primary registrar fails.

Defaults

3600 seconds

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.

Usage Guidelines

By default, SIP gateways do not generate SIP Register messages. This command enables the gateway to register E.164 telephone numbers with primary and secondary external SIP registrars.

Examples

The following is example output for the **show running-config** command.

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

Related Commands

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

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. To reset this number to the default, use the **no** form of this command.

retry register *number*

no retry register

Syntax Description

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

Defaults

10 retries

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.

Usage Guidelines

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

Examples

The following example specifies that the gateway sends nine Register messages.

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

Related Commands

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

show sip-ua register status

To display the status of E.164 numbers that a Session Initiation Protocol (SIP) gateway has registered with an external primary SIP registrar, use the **show sip-ua register status** command in privileged EXEC mode.

show sip-ua register status [**secondary**]

Syntax Description

secondary	(Optional) Displays the status of E.164 numbers that a SIP gateway registered with an external secondary SIP registrar.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.

Usage Guidelines

SIP gateways can 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.

Examples

The following is sample output from this command:

```
Router# show sip-ua register status

Line peer expires(sec) registered
4001 20001 596         no
4002 20002 596         no
5100 1     596         no
9998 2     596         no
```

Table 2 describes significant fields shown in this output.

Table 2 show sip-ua register status Field Descriptions

Field	Description
Line	The phone number to register.
peer	The registration destination number.
expires (sec)	The amount of time, in seconds, until registration expires.
registered	Registration status.

Related Commands

Command	Description
registrar	Enables SIP gateways to register E.164 numbers on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and SCCP phones with an external SIP proxy or SIP registrar.

show sip-ua statistics

To display response, traffic, and retry Session Initiation Protocol (SIP) statistics, use the **show sip-ua statistics** command in privileged EXEC mode.

show sip-ua statistics

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB	Command output was enhanced as follows: BadRequest counter (400 class) now counts malformed Via entries, reliable provisional responses (PRACK/rel1*xx*), conditions met (COMET), and NOTIFY responses.
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. For the purposes of display, this command was separated from the generic **show sip-ua** command.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T. Command output was enhanced as follows: • OkInfo counter (200 class) counts the number of successful responses to INFO requests. • Info counter counts the number of INFO messages received and sent. • BadEvent counter (489 response) counts responses to Subscribe messages with event types that are not understood by the server. • OkSubscribe counter (200 class) counts the number of 200 OK SIP messages received and sent in response to Subscribe messages. • Subscribe requests indicates total requests received and sent. • SDP application statistics added to monitor SDP. This command is supported on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 in this release.

Release	Modification
12.2(13)T	This command was integrated into Cisco IOS Release 12.2(13)T. The following cause codes have been obsoleted from the command output: • Redirection: *SeeOther* • Client Error: *LengthRequired* A new SIP statistics counter has been added: • Miscellaneous Counters: *RedirectResponseMappedToClientError* Command output was enhanced to display the following: • Time stamp that indicates the last time that SIP statistics counters were cleared.
12.2(15)ZJ	Command output was enhanced to display the following: • Register counter and statistics.

Usage Guidelines

Use the **show sip-ua statistics** command to verify SIP configurations.

Examples

The following is sample output from this command:

```
Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
    Informational:
      Trying 0/0, Ringing 0/0,
      Forwarded 0/0, Queued 0/0,
      SessionProgress 0/0
     Success:
      OkInvite 0/0, OkBye 0/0,
      OkCancel 0/0, OkOptions 0/0,
      OkPrack 0/0, OkPreconditionMet 0/0,
      OkSubscribe 0/0, OkNOTIFY 0/0,
      OkInfo 0/0, 202Accepted 0/0
      OkRegister 12/49
     Redirection (Inbound only except for MovedTemp(Inbound/Outbound)) :
      MultipleChoice 0, MovedPermanently 0,
      MovedTemporarily 0/0, UseProxy 0,
      AlternateService 0
      Client Error:
      BadRequest 0/0, Unauthorized 0/0,
      PaymentRequired 0/0, Forbidden 0/0,
      NotFound 0/0, MethodNotAllowed 0/0,
      NotAcceptable 0/0, ProxyAuthReqd 0/0,
      ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
      ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
      UnsupportedMediaType 0/0, BadExtension 0/0,
      TempNotAvailable 0/0, CallLegNonExistent 0/0,
      LoopDetected 0/0, TooManyHops 0/0,
      AddrIncomplete 0/0, Ambiguous 0/0,
      BusyHere 0/0, RequestCancel 0/0,
      NotAcceptableMedia 0/0, BadEvent 0/0,
      SETooSmall 0/0
```

```
     Server Error:
      InternalError 0/0, NotImplemented 0/0,
      BadGateway 0/0, ServiceUnavail 0/0,
      GatewayTimeout 0/0, BadSipVer 0/0,
      PreCondFailure 0/0
     Global Failure:
      BusyEverywhere 0/0, Decline 0/0,
      NotExistAnywhere 0/0, NotAcceptable 0/0
      Miscellaneous counters:
      RedirectRspMappedToClientErr 0

SIP Total Traffic Statistics (Inbound/Outbound)
      Invite 0/0, Ack 0/0, Bye 0/0,
      Cancel 0/0, Options 0/0,
      Prack 0/0, Comet 0/0,
      Subscribe 0/0, NOTIFY 0/0,
      Refer 0/0, Info 0/0
      Register 49/16

Retry Statistics
      Invite 0, Bye 0, Cancel 0, Response 0,
      Prack 0, Comet 0, Reliable1xx 0, NOTIFY 0
      Register 4

SDP application statistics:
Parses: 0, Builds 0
Invalid token order: 0, Invalid param: 0
Not SDP desc: 0, No resource: 0

Last time SIP Statistics were cleared: <never>
```

Command output, listed in Table 3, includes a reason phrase and a count describing the SIP messages received and sent. When *x/x* is included in the reason phrase field, the first number is an inbound count, and the second number is an outbound count. The description field headings are based on the SIP response code *xxx*, which the SIP protocol uses in determining behavior. SIP response codes are classified into one of the following six categories:

- 1*xx*: Informational. Indicates call progress.
- 2*xx*: Success. Indicates successful receipt or completion of a request.
- 3*xx*: Redirection. Indicates that a redirect server has returned possible locations.
- 4*xx*: Client error. Indicates that a request cannot be fulfilled as it was submitted.
- 5*xx*: Server error. Indicates that a request has failed because of an error by the server. The request may be retried at another server.
- 6*xx*: Global failure. Indicates that a request has failed and should not be tried again at any server.

Table 3 describes significant fields shown in this output, in alphabetical order.

Table 3 show sip-ua statistics Field Descriptions

Field	Description
Note For each field, the standard RFC 2543 SIP response number and message are shown.	
Ack 0/0	A confirmed final response received or sent.
Accepted 0/0	202 A successful response to a Refer request received or sent.
AddrIncomplete 0/0	484 Address supplied is incomplete.
AlternateService 0	380 Unsuccessful call; however, an alternate service is available.

Table 3 show sip-ua statistics Field Descriptions (continued)

Field	Description
Ambiguous 0/0	485 Address supplied is ambiguous.
BadEvent 0/0	489 Bad Event response indicates a Subscribe request having an event type that the server could not understand.
BadExtension 0/0	420 Server could not understand the protocol extension in the Require header.
BadGateway 0/0	502 Network is out of order.
BadRequest	400 Bad Request (includes the malformed Via header).
BadSipVer 0/0	505 Requested SIP version is not supported.
BusyEverywhere 0/0	600 Called party is busy.
BusyHere 0/0	486 Called party is busy.
Bye 0	Number of times that a Bye request is retransmitted to the other user agent.
Bye 0/0	Terminated the session.
CallLegNonExistent 0/0	481 Server is ignoring the request. Either it was a Bye request and there was no matching leg ID, or it was a Cancel request and there was no matching transaction.
Cancel 0	Number of times that a Cancel request is retransmitted to the other user agent.
Cancel 0/0	Terminated the pending request.
Comet 0	Number of times that a COMET request is retransmitted to the other user agent.
Comet 0/0	Conditions have been met.
Conflict 0/0	409 Temporary failure.
Decline 0/0	603 Call rejected.
Forbidden 0/0	403 The SIP server has the request, but cannot provide service.
Forwarded 0/0	181 Call has been forwarded.
GatewayTimeout 0/0	504 The server or gateway did not receive a timely response from another server (such as a location server).
Gone 0/0	410 Resource is no longer available at the server, and no forwarding address is known.
Info 0/0	Number of information messages that gateway has received (inbound) and how many have been transmitted (outbound).
InternalError 0/0	500 The server or gateway encountered an unexpected error that prevented it from processing the request.
Invite 0	Number of times that an Invite request is retransmitted to the other user agent.
Invite 0/0	Initiates a call.
LoopDetected 0/0	482 A loop—server received a request that included itself in the path.
MethodNotAllowed 0/0	405 Method specified in the request is not allowed.

Table 3 show sip-ua statistics Field Descriptions (continued)

Field	Description
MovedPermanently 0	301 User is no longer available at this location.
MovedTemporarily 0	302 User is temporarily unavailable.
MultipleChoice 0	300 Address resolves to more than one location.
NotAcceptable 0/0	406/606 Call was contacted, but some aspect of the session description was unacceptable.
NotAcceptableMedia 0/0	406 Call was contacted, but some aspect of the session description was unacceptable.
NotExistAnywhere 0/0	604 Server has authoritative information that the called party does not exist in the network.
NotFound 0/0	404 Called party does not exist in the specified domain.
NOTIFY 0	Number of times that a NOTIFY is retransmitted to the other user agent.
NOTIFY 0/0	Number of NOTIFY messages received or sent.
NotImplemented 0/0	501 Service or option is not implemented in the server or gateway.
OkBye 0/0	200 Successful response to a Bye request.
OkCancel 0/0	200 Successful response to a Cancel request.
OkInfo	200 Successful response to an INFO request.
OkInvite 0/0	200 Successful response to an INVITE request.
OkNOTIFY 0/0	200 Successful response to a NOTIFY request.
OkOptions 0/0	200 Successful response to an Options request.
OkPrack 0/0	200 Successful response to a PRACK request.
OkPreconditionMet 0/0	200 Successful response to a PreconditionMet request.
OkRegister 0/0	200 Successful response to a Register request.
OkSubscribe 0/0	200 Successful response to a SUBSCRIBE request.
Options 0/0	Query the receiving or sending server as to its capabilities.
PaymentRequired 0/0	402 Payment is required to complete the call.
Prack 0	Number of times that a PRACK request is retransmitted to the other user agent.
Prack 0/0	Provisional response received or sent.
PreCondFailure 0/0	580 The session could not be established because of failure to meet required preconditions.
ProxyAuthReqd 0/0	407 Rejected for proxy authentication.
Queued 0/0	182 Until the called party is available, the message is queued.

Table 3 show sip-ua statistics Field Descriptions (continued)

Field	Description
RedirectResponseMappedTo ClientError 0	Count of incoming 3*xx* responses that were mapped to 4*xx* responses. It is incremented when the **no redirection** command is active. For the default case, the 3*xx* messages are processed per RFC 2543, and this counter is not incremented. This counter counts only inbound messages and only the 3*xx* responses that are known (300, 301, 302, 305, and 380). The counter is cleared when the **clear sip-ua statistics** command is issued.
Refer 0/0	Number of Refer requests received or sent.
Register 0/0	Number of Register requests received or sent.
Register 0	Number of times that a Register request is retransmitted to the other user agent.
Reliable1xx 0	Number of times the Reliable 1*xx* response request is retransmitted to the other user agent.
ReqEntityTooLarge 0/0	413 Server refuses to process request because the request is larger than is acceptable.
ReqTimeout 0/0	408 Server could not produce a response before the Expires timeout.
RequestCancel 0/0	Request has been canceled.
ReqURITooLarge 0/0	414 Server refuses to process, because the URI (URL) request is larger than is acceptable.
Response 0	Number of Response retries.
Retry Statistics	One of the three categories of response statistics.
Ringing 0/0	180 Called party has been located and is being notified of the call.
ServiceUnavail 0/0	503 Service option is not available because of an overload or maintenance problem.
SessionProgress 0/0	183 Indicates in-band alerting.
SIP Response Statistics (Inbound/Outbound)	One of the three categories of response statistics.
SIP Total Traffic Statistics (Inbound/Outbound)	One of the three categories of response statistics.
Subscribe 0/0	Number of Subscribe requests received or sent.
TempNotAvailable 0/0	480 Called party did not respond.
TooManyHops 0/0	483 A server received a request that required more hops than is allowed by the Max-Forward header.
Trying 0/0	100 Action is being taken with no resolution.
Unauthorized 0/0	401 The request requires user authentication.
UnsupportedMediaType 0/0	415 Server refuses to process a request because the service option is not available on the destination endpoint.
UseProxy 0	305 Caller must use a proxy to contact called party.

Related Commands

Command	Description
show sip-ua retry	Displays SIP retry statistics.
show sip-ua statistics	Displays response, traffic, and retry SIP statistics.
show sip-ua timers	Displays the current settings for SIP UA timers.
sip-ua	Enables the SIP user-agent configuration commands.

show sip-ua status

To display status for the Session Initiation Protocol (SIP) user agent (UA), use the **show sip-ua status** command in privileged EXEC mode.

show sip-ua status

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.1(3)T	The statistics portion of the output was removed and included in the **show sip-ua statistics** command.
12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB	Command output was enhanced to display if media or signaling binding is enabled and the style of DNS SRV query: • 1 for RFC 2052 • 2 for RFC 2782
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. For the purposes of display, this command was separated from the generic **show sip-ua** command.
12.2(11)T	Command output was enhanced to display information on Session Description Protocol (SDP) application configuration. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.
12.2(13)T	Command output was enhanced to display the following: • Information on redirection message handling. • Information on handling of 180 responses with SDP.
12.2(15)ZJ	Command output was enhanced to display information on the duration of DTMF events.

Usage Guidelines

Use this command to verify SIP configurations.

Examples

The following is sample output from this command:

```
Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 6
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED

 SDP application configuration:
 Version line (v=) required
 Owner line (o=) required
 Session name line (s=) required
 Timespec line (t=) required
 Media supported: audio image
 Network types supported: IN
 Address types supported: IP4
 Transport types supported: RTP/AVP udptl
```

Table 4 describes significant fields shown in this output.

Table 4 show sip-ua status Field Descriptions

Field	Description
SIP User Agent Status	UA status.
SIP User Agent for UDP	UDP is enabled or disabled.
SIP User Agent for TCP	TCP is enabled or disabled.
SIP User Agent bind status (signaling)	Binding for signaling is enabled or disabled.
SIP User Agent bind status (media)	Binding for media is enabled or disabled.
SIP early-media for 180 responses with SDP	Early media cut-through treatment for 180 responses with SDP can be enabled, the default treatment, or disabled, with local ringback provided.
SIP max-forwards	Value of max-forwards of SIP messages.
SIP DNS SRV version	Style of DNS SRV query: 1 for RFC 2052 or 2 for RFC 2782.
NAT Settings for the SIP-UA	Symmetric Network Address Traversal (NAT) settings when the feature is enabled.
Role in SDP	Identifies the endpoint function in the connection setup procedure during symmetric NAT traversal. The endpoint role may be set to active, meaning it initiates a connection, or passive, meaning it accepts a connection. A value of none in this field means the feature is disabled.
Check media source packets	Media source packet checking is enabled or disabled.

Table 4 show sip-ua status Field Descriptions (continued)

Field	Description
Maximum duration for a telephone-event in NOTIFYs	Shows the time interval between consecutive NOTIFY messages for a telephone event.
SIP support for ISDN SUSPEND/RESUME	Suspend and Resume support is enabled or disabled.
Redirection (3*xx*) message handling	Redirection can be enabled, which is the default status as per RFC 2543. Or handling of redirection 3*xx* messages can be disabled, allowing the gateway to treat 3*xx* redirect messages as 4*xx* error messages.
Version	SDP version.
Owner	Session originator.
Session name	Identifies the session name.
Timespec	Identifies the session start and stop times.
Media supported	Media information.
Network types supported	Always IN for Internet.
Address types supported	Identifies the Internet Protocol version.
Transport type supported	Identifies the transport protocols supported.

Related Commands

Command	Description
show sip-ua retry	Displays SIP retry statistics.
show sip-ua statistics	Displays response, traffic, and retry SIP statistics.
show sip-ua timers	Displays the current settings for SIP UA timers.
sip-ua	Enables the SIP user-agent configuration commands.

show sip-ua timers

To display the current settings for the Session Initiation Protocol (SIP) user-agent (UA) timers, use the **show sip-ua timers** command in privileged EXEC mode.

show sip-ua timers

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.1(3)T	The output of this command was changed to reflect the changes in the **timers** command.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB	Command output was enhanced to display the following: Reliable provisional responses (PRACK/rel 1*xx*), Conditions met (COMET), and NOTIFY responses.
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. For the purposes of display, this command was separated from the generic **show sip-ua** command found previously in this reference.
12.2(11)T	This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.
12.2(11)YT	Command output was enhanced to display Refer responses.
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.
12.2(15)ZJ	Command output was enhanced to display Register responses.

Usage Guidelines

Use this command to verify SIP configurations.

Examples

The following is sample output from this command:

```
Router# show sip-ua timers

SIP UA Timer Values (millisecs)
trying 500, expires 180000, connect 500, disconnect 500
comet 500, prack 500, rel1xx 500, notify 500
refer 500, register 500
```

Table 5 describes significant fields shown in this output.

Table 5 show sip-ua timers Field Descriptions

Field	Description
SIP UA Timer Values (millisecs)	SIP UA timer status.
trying	Time to wait before a Trying message is retransmitted.
expires	Time to wait before an Expires message is retransmitted.
connect	Time to wait before a Connect message is retransmitted.
disconnect	Time to wait before a Disconnect message is retransmitted.
comet	Time to wait before a COMET message is retransmitted.
prack	Time to wait before a PRACK acknowledgment is retransmitted.
rel1xx	Time to wait before a Rel1*xx* response is retransmitted.
notify	Time to wait before a NOTIFY response is retransmitted.
refer	Time to wait before a Retry request is retransmitted.
register	Time to wait before a Register request is retransmitted.

Related Commands

Command	Description
show sip-ua retry	Displays SIP retry statistics.
show sip-ua statistics	Displays response, traffic, and retry SIP statistics.
show sip-ua status	Displays SIP UA status.
sip-ua	Enables the SIP user-agent configuration commands.

timers register

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

timers register *time*

no timers register

Syntax Description

time	Waiting time, in milliseconds. Range is from 100 to 1000. Default is 500.

Defaults

500 milliseconds

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.

Examples

The following example sends register requests every 500 milliseconds:

```
sip-ua
 retry invite 9
 retry register 9
 timers register 500
```

Related Commands

Command	Description
retry register	Sets the total number of SIP registers to send.

Glossary

call—In SIP, a call consists of all participants in a conference who are invited by a common source. A SIP call is identified by a globally unique call identifier. A point-to-point IP telephony conversation maps into a single SIP call.

DNS—Domain Name System. Used to translate H.323 IDs, URLs, or e-mail IDs to IP addresses. DNS is also used to assist in locating remote gatekeepers and to reverse-map raw IP addresses to host names of administrative domains.

DNS SRV—Domain Name System Server. Used to locate servers for a given service.

DTMF—dual-tone multifrequency. Tones that are generated when a button on a touch-tone phone is pressed. When the tone is generated, it is compressed, transported to the other party, and decompressed.

DTMF relay— DTMF relay provides reliable digit relay between VoIP gateways when a low-bandwidth codec is used. DTMF relay provides a standardized means of transporting DTMF tones in Real-Time Transport Protocol (RTP) packets and is identified by dynamic payload types in the SDP.

INVITE—A SIP message that initiates a SIP session. It indicates that a user is invited to participate, provides a session description, indicates the type of media, and provides information regarding the capabilities of the called and calling parties.

NOTIFY—SIP NOTIFY messages report when certain events occur, such as DTMF events.

proxy—A SIP UAC or UAS that forwards requests and responses on behalf of another SIP UAC or UAS.

PSTN—public switched telephone network. PSTN refers to the local telephone company network.

SCCP—Skinny Client Control Protocol. SCCP is the Cisco standard for real-time calls and conferencing over IP. This generalized messaging set allows Cisco IP phones to coexist in an H.323 environment. The savings in memory size, processor power, and complexity makes the protocol desirable.

session—A SIP session includes a set of multimedia senders and receivers and the data streams that flow between the senders and receivers. A SIP multimedia conference is an example of a session. The called party can be invited several times by different calls to the same session.

SIP—Session Initiation Protocol. An application-layer protocol originally developed by the Multiparty Multimedia Session Control (MMUSIC) working group of the Internet Engineering Task Force (IETF). Their goal was to equip platforms to signal the setup of voice and multimedia calls over IP networks. SIP features are compliant with IETF RFC 2543, published in March 1999.

SRST—Survivable Remote Site Telephony.

URI—Uniform Resource Identifier. Takes a form similar to an e-mail address, indicates the user's SIP identity, and is used for redirection of SIP messages.

URL—Uniform Resource Locator. Standard address of any resource on the Internet that is part of the World Wide Web (WWW).

VoIP—Voice over IP. The ability to carry normal telephone-style voice over an IP-based network.

Note Refer to the *Internetworking Terms and Acronyms* for terms not included in this glossary.