SIP: Connection-Oriented Media Enhancements for SIP

The Connection-Oriented Media (Comedia) Enhancements for SIP feature allows the Cisco gateway to check the media source of incoming Realtime Transport Protocol (RTP) packets, and allows the endpoint to advertise its presence inside or outside of Network Address Translation (NAT). Using the new feature enables symmetric NAT traversal by supporting the capability to modify and update an existing RTP session remote address and port.

Feature Specifications for Connection-Oriented Media (Comedia) Enhancements for SIP

Feature History	
Release	**Modification**
12.2(13)T	This feature was introduced.
Supported Platforms	
Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850, and Cisco CVA 120 series platforms.	

Availability of Cisco IOS Software Images

Platform support for particular Cisco IOS software releases is dependent on the availability of the software images for those platforms. Software images for some platforms may be deferred, delayed, or changed without prior notice. For updated information about platform support and availability of software images for each Cisco IOS software release, refer to the online release notes or, if supported, Cisco Feature Navigator.

Contents

Prerequisites for Connection-Oriented Media (Comedia) Enhancements for SIP

- Ensure that your Cisco router has the minimum memory requirements necessary for voice capabilities.
- Ensure that the gateway has voice functionality configured for SIP.

  For more information about configuring SIP, refer to

  Cisco IOS Voice, Video, and Fax Configuration Guide, Release 12.2, the "Configuring SIP for VoIP" chapter.
- Establish a working IP network.

  For more information about configuring IP, refer to

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

  For more information about configuring NAT, refer to:

  Configuring Network Address Translation: Getting Started.

Restrictions for Connection-Oriented Media (Comedia) Enhancements for SIP

The new feature does not support the a=direction:both attribute of the Session Description Protocol (SDP) message**,** as defined in the Internet Engineering Task Force (IETF) draft, draft-ietf-mmusic-sdp-comedia-04.txt, *Connection-Oriented Media Transport in SDP.* There is likewise no corresponding command-line interface (CLI) command. If the SIP gateway receives an SDP message specifying a=direction:both, the endpoint is treated by the gateway as active, and considered to be inside the NAT.

Note Proxy parallel forking is not supported with this feature unless all endpoints reply with 180 message response without SDP, because this feature does not handle media coming from multiple endpoints simultaneously.

Information About Connection-Oriented Media (Comedia) Enhancements for SIP

To configure the Connection-Oriented Media (Comedia) Enhancements for SIP feature, you must understand the following concepts:

Benefits of Connection-Oriented Media (Comedia) Enhancements for SIP

- Ability to check the media source address and port of incoming RTP packets, thereby enabling the remote address and port of the existing session to be updated
- Enhanced interoperability in networks where NAT devices are unaware of SIP or SDP signaling
- Ability to advertise endpoint presence inside or outside NAT
- Ability to specify the connection role of the endpoint

Symmetric NAT Traversal

The Connection-Oriented Media (Comedia) Enhancements for SIP feature provides the following functionality to symmetric NAT traversal:

- Allows the Cisco gateway to check the media source of incoming (RTP) packets.
- Allows the endpoint to advertise its presence inside or outside of NAT.

NAT, which maps the source IP address of a packet from one IP address to a different IP address, has varying functionality and configurations. NAT can help conserve IP version 4 (IPv4) addresses, or it can be used for security purposes to hide the IP address and LAN structure behind the NAT. VoIP endpoints may both be outside NAT, both inside, or one inside and the other outside.

In symmetric NAT, all requests from an internal IP address and port to a specific destination IP address and port are mapped to the same external IP address and port. The new feature provides additional capabilities for symmetric NAT traversal.

Prior to the implementation of connection-oriented media enhancements, NAT traversal presented challenges for both SIP, which signals the protocol messages that set up a call, and for RTP, the media stream that transports the audio portion of a VoIP call. An endpoint with connections to clients behind NATs and on the open Internet had no way of knowing when to trust the addressing information it received in the SDP portion of SIP messages, or whether to wait until it received a packet directly from the client before opening a channel back to the source IP:port of that packet. Once a VoIP session was established, the endpoint was, in some scenarios, sending packets to an unreachable address. This scenario typically occurred in NAT networks that were SIP-unaware.

In addition to the challenges posed by NAT traversal in SIP, NAT traversal in RTP requires that a client must know what type of NAT it sits behind, and that it must also obtain the public address for an RTP stream. Any RTP connection between endpoints outside and inside NAT must be established as a point-to-point connection. The external endpoint must wait until it receives a packet from the client so that it knows where to reply. The connection-oriented protocol used to describe this type of session is known as Connection-Oriented Media (Comedia), as defined in the IETF draft, draft-ietf-mmusic-sdp-comedia-04.txt, *Connection-Oriented Media Transport in SDP.*

The Connection-Oriented Media (Comedia) Enhancements for SIP feature implements one of many possible SIP solutions to address problems with different NAT types and traversals. With the new feature the gateway can open an RTP session with the remote end and then update or modify the existing RTP session remote address and port (raddr:rport) with the source address and port of the actual media packet received after passing through NAT. The new feature allows you to configure the gateway to modify the RTP session remote address and port by implementing support for the SDP direction (a=direction:<role>) attribute defined in, draft-ietf-mmusic-sdp-comedia-04.txt*, Connection-Oriented Media Transport in SDP.* Valid values for the attribute are as follows:

- active, which indicates that the endpoint initiates a connection to the port number on the m= line of the session description from the other endpoint.
- passive, which indicates that the endpoint accepts a connection to the port number on the m= line of the session description sent from itself to the other endpoint.
- both, which indicates the that endpoint both accepts an incoming connection and initiates an outgoing connection to the port number on the m= line of the session description from the other endpoint.

The new feature introduces new CLI commands to configure the following SIP user agent settings for symmetric NAT:

- The **nat symmetric check-media-src** command enables checking the incoming packet for media source address. This capability allows the gateway to check the source address and update the media session with the new remote media address and port.
- The **nat symmetric role** command specifies the function of the endpoint in the connection setup procedure. The **role** keyword may be set to one of the following:
  - **active**, meaning the endpoint initiates a connection to the port number on the m= line of the session description from the other endpoint.
  - **passive**, meaning the endpoint accepts a connection to the port number on the m= line of the session description sent from itself to the other endpoint.

Note The Cisco comedia implementation does not support a=direction:both. If the Cisco gateway receives a=direction:both in the SDP message, the endpoint is considered active.

Sample SDP Message

The following example shows a sample SDP message that describes a session with the direction:<role> attribute set to passive:

```
v=o
o=CiscoSystemsSIP-GW-UserAgent 5732 7442 IN IP4 10.15.66.43
s=SIP Call
c=IN IP4 10.15.66.43
t=0 0
m=audio 17306 RTP/AVP 0 100
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=ptime:20
a=direction:passive
```

Symmetric NAT Call Flows

The following call flow diagrams describe call setup during symmetric NAT traversal scenarios. Figure 1 shows a NAT device that is unaware of SIP or SDP signaling. The SIP endpoints are not communicating the connection-oriented media direction role in the SDP message. The originating gateway is configured, using the command **nat symmetric check-media-src,** to detect the media source and update the VoIP RTP session to the network address translated address:port pair.

Figure 1 SIP Endpoints Not Communicating the SDP direction:<role> Attribute

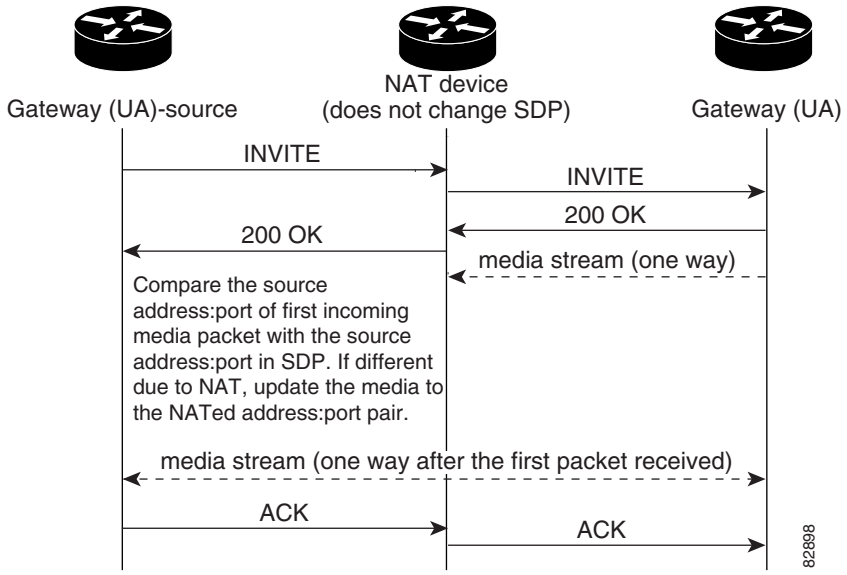
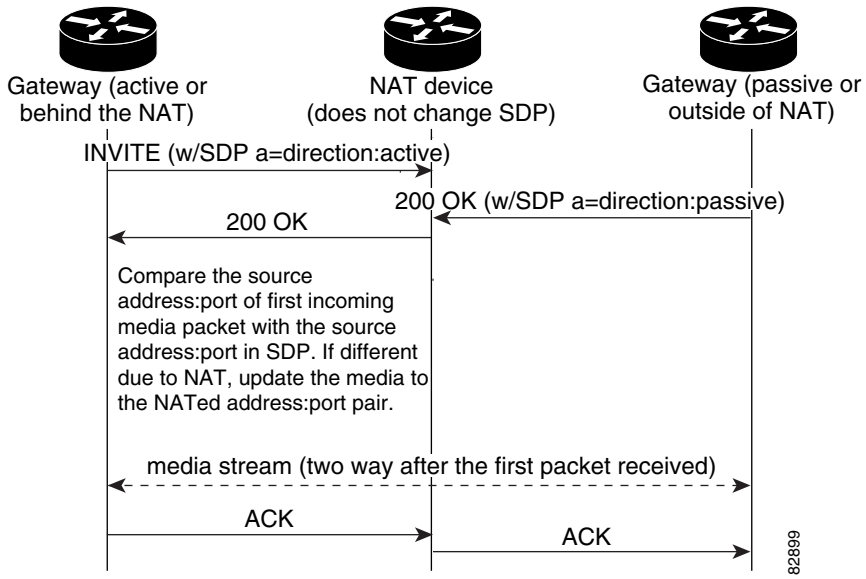


Figure 2 shows a NAT device that is unaware of SIP or SDP signaling, but the SIP endpoints are communicating the connection-oriented media direction role in the SDP message. The originating gateway is configured as a passive entity in the network using the **nat symmetric role** command. When the passive entity receives a direction role of active, it updates the VoIP RTP session to the network address translated address:port pair.

Figure 2 SIP Endpoints Communicating the SDP direction:<role>



Note Configuring the originating gateway for passive or active setting can differ from the NAT device setup. The SIP user agent communicates the CLI configured direction role in the SDP body. Checking for media packets is automatically enabled only if the gateway receives a direction role of active or both.

How to Configure Comedia Enhancements for SIP

This section contains the following procedures. Each procedure is identified as either required or optional.

- Configuring Connection-Oriented Media (Comedia) Enhancements for SIP, page 7 (required)
- Verifying Connection-Oriented Media (Comedia) Enhancements for SIP, page 8 (optional)

Configuring Connection-Oriented Media (Comedia) Enhancements for SIP

Perform the following tasks to enable the gateway to check the media source address and port of the first incoming RTP packet, and to optionally specify whether the endpoint is active or passive. Once the media source check is enabled, the gateway can modify or update the established VoIP RTP session with upstream addressing information extracted from the SDP body of the received request.

SUMMARY STEPS

1. **enable**
2. **configure** [**terminal** | **memory** | **network**]
3. **sip-ua**
4. **nat symmetric check-media-source**
5. **nat symmetric role** {**active** | **passive**}
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure** {**terminal** | **memory** | **network**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	`Router(config# sip-ua` **Example:** `Router(config# sip-ua`	Enters SIP-UA configuration mode.
Step 4	**nat symmetric** c**heck-media-source** **Example:** `Router(config-sip-ua)# nat symmetric check-media-source`	(Required) Specifies settings for the SIP user agent in symmetric NAT configuration. • In this example, the gateway is configured to perform source media checking for symmetric NAT.
Step 5	**nat symmetric role** {**active** | **passive**} **Example:** `Router(config-sip-ua)# nat symmetric role active`	(Optional) Specifies endpoint settings for the SIP user agent in symmetric NAT configuration. The default setting is passive. • The optional **active** keyword configures the endpoint ability to initiate an outgoing connection. • The optional **passive** keyword configures the endpoint ability to accept an incoming connection.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user agent configuration mode

Verifying Connection-Oriented Media (Comedia) Enhancements for SIP

Perform this task to verify that the Connection-Oriented Media (Comedia) Enhancements for SIP feature is working.

SUMMARY STEPS

1. **enable**
2. **show running-config**
3. **debug ccsip all**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**show running-config** **Example:** `Router# show running-config`	(Optional) Displays the configuration information currently running on the router. View the sip-ua configuration to verify Comedia settings.
Step 3	**debug ccsip all** **Example:** `Router# debug ccsip all`	(Optional) Enables all SIP call tracing. View the direction attribute settings and port and network address translation traces to verify Comedia configuration.

Configuration Examples for Connection-Oriented Media (Comedia) Enhancements for SIP

This section provides the following configuration examples:

Configuring Source Media Check Example

The following example shows how to enable checking the media source address and port of incoming RTP packets:

```
Router(config)# sip-ua
Router(config-sip-ua)# nat symmetric check-media-src
```

Configuring the Endpoint Connection Role Example

The following example shows how to configure the endpoint role in connection setup to passive:

```
Router(config)# sip-ua
Router(config-sip-ua)# nat symmetric role passive
```

Verifying Connection-Oriented Media (Comedia) Enhancements for SIP Examples

In the following examples, the output is displayed for each command used in the section "Verifying Connection-Oriented Media (Comedia) Enhancements for SIP."

Sample Output for the show running-config Command

You can use the **show running-config** command to verify that source media checking is enabled:

```
Router# show running-config
Building configuration...

Current configuration :2791 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 3640
!
voice-card 2
!
ip subnet-zero
!
no ip domain lookup
ip domain name cisco.com
ip name-server 172.18.195.113
!
isdn switch-type primary-ni
!
fax interface-type fax-mail
mta receive maximum-recipients 0
ccm-manager mgcp
!
controller T1 2/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
controller T1 2/1
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
interface Ethernet0/0
 ip address 172.18.197.22 255.255.255.0
 half-duplex
!
interface Serial0/0
 no ip address
 shutdown
!
interface TokenRing0/0
 no ip address
 shutdown
 ring-speed 16
!
interface FastEthernet1/0
 no ip address
 shutdown
 duplex auto
```

```
 speed auto
!
interface Serial2/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn outgoing display-ie
 no cdp enable
!
interface Serial2/1:23
 no ip address
 no logging event link-status
 isdn switch-type primary-ni
 isdn incoming-voice voice
 isdn outgoing display-ie
 no cdp enable
!
ip classless
ip route 0.0.0.0 0.0.0.0 Ethernet0/0
no ip http server
ip pim bidir-enable
!
call rsvp-sync
!
voice-port 2/0:23
!
voice-port 2/1:23
!
voice-port 3/0/0
!
voice-port 3/0/1
!
mgcp ip qos dscp cs5 media
mgcp ip qos dscp cs3 signaling
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 646 voip
destination-pattern 5552222
 session protocol sipv2
 session target ipv4:10.0.0.1
!
dial-peer voice 700 pots
 destination-pattern 700#T
port 0:D
!
gateway
!
sip-ua
    nat symmetric check-media-src
    max-forwards 5
!
line con 0
line aux 0
line vty 0 4
 login
!
end
```

Sample Output for the debug ccsip all Command

In the following example, output is displayed with the **role** keyword of the **nat symmetric** command set to **active** for the originating gateway, and to **passive** for the terminating gateway.

```
Router3640# debug ccsip all
All SIP call tracing enabled
Router3640#
00:02:12:0x6327E424 :State change from (UNDEFINED, SUBSTATE_NONE)  to (STATE_IDLE,
SUBSTATE_NONE)
00:02:12:****Adding to UAC table

00:02:12:adding call id 3 to table

00:02:12:Queued event from SIP SPI :SIPSPI_EV_CC_CALL_SETUP (10)
00:02:12:CCSIP-SPI-CONTROL: act_idle_call_setup
00:02:12: act_idle_call_setup:Not using Voice Class Codec

00:02:12:act_idle_call_setup:preferred_codec set[0] type :g711ulaw bytes:160
00:02:12:sipSPICopyPeerDataToCCB:From CLI:Modem NSE payload = 100, Passthrough = 0,Modem
relay = 0, Gw-Xid = 1
SPRT latency 200, SPRT Retries = 12, Dict Size = 1024
 String Len = 32, Compress dir = 3
00:02:12:****Deleting from UAC table

00:02:12:****Adding to UAC table

00:02:12:Queued event from SIP SPI :SIPSPI_EV_CREATE_CONNECTION (6)
00:02:12:0x6327E424 :State change from (STATE_IDLE, SUBSTATE_NONE)  to (STATE_IDLE,
SUBSTATE_CONNECTING)
00:02:12:0x6327E424 :State change from (STATE_IDLE, SUBSTATE_CONNECTING)  to (STATE_IDLE,
SUBSTATE_CONNECTING)
00:02:12:sipSPIUsetBillingProfile:sipCallId for billing records =
D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
00:02:12:CCSIP-SPI-CONTROL: act_idle_connection_created
00:02:12:CCSIP-SPI-CONTROL: act_idle_connection_created:Connid(1) created to
172.18.200.237:5060, local_port 56992
00:02:12:CCSIP-SPI-CONTROL: sipSPIOutgoingCallSDP
00:02:12: Preferred method of dtmf relay is:6, with payload :101

00:02:12: convert_codec_bytes_to_ptime:Values :Codec:g711ulaw codecbytes :160, ptime:20

00:02:12:sip_generate_sdp_xcaps_list:Modem Relay disabled. X-cap not needed

00:02:12:CCSIP-SPI-CONTROL: Clock Time Zone is UTC, same as GMT:Using GMT
00:02:12:sipSPIAddLocalContact
00:02:12:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
00:02:12:sip_stats_method
00:02:12:0x6327E424 :State change from (STATE_IDLE, SUBSTATE_CONNECTING)  to
(STATE_SENT_INVITE, SUBSTATE_NONE)
00:02:12:Sent:
INVITE sip:2021010124@172.18.200.237:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>
Date:Mon, 01 Mar 1993 00:02:12 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Supported:timer,100rel
Min-SE: 1800
Cisco-Guid:3563045876-351146444-2147852364-2382746380
User-Agent:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Max-Forwards:1
Timestamp:730944132
Contact:<sip:888001@10.15.66.43:5060;user=phone>
Expires:60
```

```
Allow-Events:telephone-event
Content-Type:application/sdp
Content-Length:291

v=0
o=CiscoSystemsSIP-GW-UserAgent 9502 9606 IN IP4 10.15.66.43
s=SIP Call
c=IN IP4 10.15.66.43
t=0 0
m=audio 16398 RTP/AVP 0 100 101
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=direction:active

00:02:12:CCSIP-SPI-CONTROL: act_sentinvite_wait_100
00:02:12:CCSIP-SPI-CONTROL: Clock Time Zone is UTC, same as GMT:Using GMT
00:02:12:sipSPIAddLocalContact
00:02:12:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
00:02:12:sip_stats_method
00:02:12:Sent:
INVITE sip:2021010124@172.18.200.237:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>
Date:Mon, 01 Mar 1993 00:02:12 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Supported:timer,100rel
Min-SE: 1800
Cisco-Guid:3563045876-351146444-2147852364-2382746380
User-Agent:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Max-Forwards:1
Timestamp:730944132
Contact:<sip:888001@10.15.66.43:5060;user=phone>
Expires:60
Allow-Events:telephone-event
Content-Type:application/sdp
Content-Length:291

v=0
o=CiscoSystemsSIP-GW-UserAgent 9502 9606 IN IP4 10.15.66.43
s=SIP Call
c=IN IP4 10.15.66.43
t=0 0
m=audio 16398 RTP/AVP 0 100 101
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=direction:active

00:02:12:Received:
SIP/2.0 100 Trying
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:57:53 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
```

```
Timestamp:730944132
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Allow-Events:telephone-event
Content-Length:0

00:02:12:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:172.18.200.237:5060
00:02:12:CCSIP-SPI-CONTROL: act_sentinvite_new_message
00:02:12:CCSIP-SPI-CONTROL: sipSPICheckResponse
00:02:12:sip_stats_status_code
00:02:12: Roundtrip delay 32 milliseconds for method INVITE

00:02:12:0x6327E424 :State change from (STATE_SENT_INVITE, SUBSTATE_NONE)  to
(STATE_RECD_PROCEEDING, SUBSTATE_PROCEEDING_PROCEEDING)
00:02:13:Received:
SIP/2.0 183 Session Progress
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:57:53 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Timestamp:730944132
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Require:100rel
RSeq:5975
Allow-Events:telephone-event
Contact:<sip:2021010124@172.18.200.237:5060;user=phone>
Content-Type:application/sdp
Content-Disposition:session;handling=required
Content-Length:240

v=0
o=CiscoSystemsSIP-GW-UserAgent 1692 40 IN IP4 172.18.200.237
s=SIP Call
c=IN IP4 172.18.200.237
t=0 0
m=audio 16898 RTP/AVP 0 100
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=ptime:20
a=direction:passive

00:02:13:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:172.18.200.237:5060
00:02:13:CCSIP-SPI-CONTROL: act_recdproc_new_message
00:02:13:CCSIP-SPI-CONTROL: sipSPICheckResponse
00:02:13:sip_stats_status_code
00:02:13: Roundtrip delay 708 milliseconds for method INVITE

00:02:13:sipSPIGetSdpBody :Parse incoming session description
00:02:13:HandleSIP1xxSessionProgress:Content-Disposition received in 18x
response:session;handling=required
00:02:13:sipSPIDoFaxMediaNegotiation()
00:02:13:sipSPIDoMediaNegotiation:Codec (g711ulaw) Negotiation Successful on Static
Payload

00:02:13: sipSPIDoPtimeNegotiation:One ptime attribute found - value:20
00:02:13: convert_ptime_to_codec_bytes:Values :Codec:g711ulaw ptime :20, codecbytes:160

00:02:13: convert_codec_bytes_to_ptime:Values :Codec:g711ulaw codecbytes :160, ptime:20

00:02:13: Parsed the direction:role identified as:0
```

```
00:02:13:sipSPIDoDTMFRelayNegotiation:Requested DTMF-RELAY option(s) not found in
Preferred DTMF-RELAY option list!
00:02:13: sipSPIDoMediaNegotiation:DTMF Relay mode :Inband Voice

00:02:13:sip_sdp_get_modem_relay_cap_params:
00:02:13:sip_sdp_get_modem_relay_cap_params:NSE payload from X-cap = 0
00:02:13:sip_do_nse_negotiation:NSE Payload 100 found in SDP
00:02:13:sip_do_nse_negotiation:Remote NSE payload = local one = 100, Use it
00:02:13:sip_select_modem_relay_params:X-tmr not present in SDP. Disable modem relay
00:02:13:sipSPIDoQoSNegotiation - SDP body with media description
00:02:13:sipSPIUpdCcbWithSdpInfo:SDP Media Information:
Negotiated Codec      :g711ulaw , bytes :160
Early Media           :0
Delayed Media         :0
Bridge Done           :0
New Media             :0
DSP DNLD Reqd         :0
Media Dest addr/Port  :172.18.200.237:16898
Orig Media Addr/Port  :0.0.0.0:0

00:02:13:0x6327E424 :State change from (STATE_RECD_PROCEEDING,
SUBSTATE_PROCEEDING_PROCEEDING)  to (STATE_RECD_PROCEEDING, SUBSTATE_PROCEEDING_PROGRESS)
00:02:13:ccsip_process_response_contact_record_route
00:02:13:0x6327E424 :State change from (STATE_RECD_PROCEEDING,
SUBSTATE_PROCEEDING_PROGRESS)  to (STATE_RECD_PROCEEDING, SUBSTATE_CONNECTING)
00:02:13:Queued event from SIP SPI :SIPSPI_EV_CREATE_CONNECTION (6)
00:02:13:0x6327E424 :State change from (STATE_RECD_PROCEEDING, SUBSTATE_CONNECTING)  to
(STATE_RECD_PROCEEDING, SUBSTATE_CONNECTING)
00:02:13:sipSPIRtcpUpdates:rtcp_session info
               laddr = 10.15.66.43, lport = 16398, raddr = 172.18.200.237, rport=16898
00:02:13:sipSPIRtcpUpdates:NO extraction of source address from remote media

00:02:13: sipSPIRtcpUpdates No rtp session in bridge, create a new one

00:02:13:CCSIP-SPI-CONTROL: ccsip_caps_ind
00:02:13:ccsip_get_rtcp_session_parameters:CURRENT VALUES:
ccCallID=3, current_seq_num=0x1500
00:02:13:ccsip_get_rtcp_session_parameters:NEW VALUES:
ccCallID=3, current_seq_num=0xB93
00:02:13:ccsip_caps_ind:Load DSP with negotiated codec :g711ulaw, Bytes=160
00:02:13:sipSPISetDTMFRelayMode:set DSP for dtmf-relay =
CC_CAP_DTMF_RELAY_INBAND_VOICE_AND_OOB
00:02:13:sip_set_modem_caps:Negotiation already Done. Set negotiated Modem caps
00:02:13:sip_set_modem_caps:Modem Relay & Passthru both disabled
00:02:13:sip_set_modem_caps:nse payload = 100, ptru mode = 0, ptru-codec=0, redundancy=0,
xid=0, relay=0, sprt-retry=12, latecncy=200, compres-dir=3, dict=1024, strnlen=32
00:02:13:ccsip_caps_ind:Load DSP with codec :g711ulaw, Bytes=160
00:02:13:CCSIP-SPI-CONTROL: ccsip_caps_ack
00:02:13:CCSIP-SPI-CONTROL: act_recdproc_connection_created
00:02:13:CCSIP-SPI-CONTROL: sipSPICheckSocketConnection:Connid(2) created to
172.18.200.237:5060, local_port 50689
00:02:13:0x6327E424 :State change from (STATE_RECD_PROCEEDING, SUBSTATE_CONNECTING)  to
(STATE_RECD_PROCEEDING, SUBSTATE_NONE)
00:02:13:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
00:02:13:sip_stats_method
00:02:13:0x6327E424 :State change from (STATE_RECD_PROCEEDING, SUBSTATE_NONE)  to
(STATE_RECD_PROCEEDING, SUBSTATE_PROCEEDING_PROGRESS)
00:02:13:Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:57:53 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
```

```
Timestamp:730944132
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Allow-Events:telephone-event
Contact:<sip:2021010124@172.18.200.237:5060;user=phone>
Content-Type:application/sdp
Content-Length:240

v=0
o=CiscoSystemsSIP-GW-UserAgent 1692 40 IN IP4 172.18.200.237
s=SIP Call
c=IN IP4 172.18.200.237
t=0 0
m=audio 16898 RTP/AVP 0 100
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=ptime:20
a=direction:passive

00:02:13:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:172.18.200.237:5060
00:02:13:Sent:
PRACK sip:2021010124@172.18.200.237:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Mon, 01 Mar 1993 00:02:12 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
CSeq:102 PRACK
RAck:5975 101 INVITE
Content-Length:0

00:02:13:CCSIP-SPI-CONTROL: act_recdproc_new_message
00:02:13:CCSIP-SPI-CONTROL: sipSPICheckResponse
00:02:13:sip_stats_status_code
00:02:13: Roundtrip delay 736 milliseconds for method PRACK

00:02:13:sipSPIGetSdpBody :Parse incoming session description
00:02:13:CCSIP-SPI-CONTROL: sipSPIUACSessionTimer
00:02:13:CCSIP-SPI-CONTROL: act_recdproc_continue_200_processing
00:02:13:CCSIP-SPI-CONTROL: act_recdproc_continue_200_processing:*** This ccb is the
parent

00:02:13:sipSPIDoFaxMediaNegotiation()
00:02:13:sipSPIDoMediaNegotiation:Codec (g711ulaw) Negotiation Successful on Static
Payload

00:02:13: sipSPIDoPtimeNegotiation:One ptime attribute found - value:20
00:02:13: convert_ptime_to_codec_bytes:Values :Codec:g711ulaw ptime :20, codecbytes:160

00:02:13: convert_codec_bytes_to_ptime:Values :Codec:g711ulaw codecbytes :160, ptime:20

00:02:13: Parsed the direction:role identified as:0

00:02:13:sipSPIDoDTMFRelayNegotiation:Requested DTMF-RELAY option(s) not found in
Preferred DTMF-RELAY option list!
00:02:13: sipSPIDoMediaNegotiation:DTMF Relay mode :Inband Voice

00:02:13:sip_sdp_get_modem_relay_cap_params:
00:02:13:sip_sdp_get_modem_relay_cap_params:NSE payload from X-cap = 0
00:02:13:sip_do_nse_negotiation:NSE Payload 100 found in SDP
00:02:13:sip_do_nse_negotiation:Remote NSE payload = local one = 100, Use it
00:02:13:sip_select_modem_relay_params:X-tmr not present in SDP. Disable modem relay
00:02:13: sipSPICompareSDP:Flags set:NEW_MEDIA :0 DSPDNLD REQD:0
```

```
00:02:13:sipSPIUpdCcbWithSdpInfo Bridge was done and there are no fqdn queries in
progress, do RTCP updates

00:02:13:sipSPIRtcpUpdates:rtcp_session info
               laddr = 10.15.66.43, lport = 16398, raddr = 172.18.200.237, rport=16898
00:02:13:sipSPIRtcpUpdates:NO extraction of source address from remote media

00:02:13: sipSPIRtcpUpdates rtp session already created in bridge - update

00:02:13:sipSPIUpdCcbWithSdpInfo:SDP Media Information:
Negotiated Codec      :g711ulaw , bytes :160
Early Media           :0
Delayed Media         :0
Bridge Done           :1048576
New Media             :0
DSP DNLD Reqd         :0
Media Dest addr/Port  :172.18.200.237:16898
Orig Media Addr/Port  :0.0.0.0:0

00:02:13:sipSPIProcessMediaChanges
00:02:13:ccsip_process_response_contact_record_route
00:02:13:CCSIP-SPI-CONTROL: sipSPIProcess200OKforinvite
00:02:13:RequestCloseConnection:Closing connid 1 Local Port 50689
00:02:13:Queued event from SIP SPI :SIPSPI_EV_CLOSE_CONNECTION (8)
00:02:13:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
00:02:13:sip_stats_method
00:02:13:0x6327E424 :State change from (STATE_RECD_PROCEEDING,
SUBSTATE_PROCEEDING_PROGRESS)  to (STATE_ACTIVE, SUBSTATE_NONE)
00:02:13:The Call Setup Information is :
Call Control Block (CCB) :0x6327E424
State of The Call        :STATE_ACTIVE
TCP Sockets Used         :NO
Calling Number           :888001
Called Number            :2021010124
Negotiated Codec         :g711ulaw
Negotiated Codec Bytes   :160
Negotiated Dtmf-relay    :0
Dtmf-relay Payload       :0

00:02:13:
 Source IP Address (Sig  ):10.15.66.43
Source IP Address (Media):10.15.66.43
Source IP Port    (Media):16398
Destn  IP Address (Media):172.18.200.237
Destn  IP Port    (Media):16898
Destn SIP Req Addr:Port  :172.18.200.237:5060
Destn SIP Resp Addr:Port :0.0.0.0:0
Destination Name         :172.18.200.237

00:02:13:
 Orig Destn IP Address:Port (Media):0.0.0.0:0

00:02:13:udpsock_close_connect:Socket fd:1 closed for connid 1 with remote port:5060
00:02:13:Sent:
ACK sip:2021010124@172.18.200.237:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Mon, 01 Mar 1993 00:02:12 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Max-Forwards:1
Content-Length:0
CSeq:101 ACK
```

```
00:02:13:Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:57:54 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Server:Cisco-SIPGateway/IOS-12.x
CSeq:102 PRACK
Content-Length:0

00:02:13:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:172.18.200.237:5060
00:02:13:CCSIP-SPI-CONTROL: act_active_new_message
00:02:13:CCSIP-SPI-CONTROL: sact_active_new_message_response
00:02:13:CCSIP-SPI-CONTROL: sipSPICheckResponse
00:02:27:Queued event from SIP SPI :SIPSPI_EV_CC_CALL_DISCONNECT (15)
00:02:27:CCSIP-SPI-CONTROL: act_active_disconnect
00:02:27:RequestCloseConnection:Closing connid 2 Local Port 50689
00:02:27:Queued event from SIP SPI :SIPSPI_EV_CLOSE_CONNECTION (8)
00:02:27:Queued event from SIP SPI :SIPSPI_EV_CREATE_CONNECTION (6)
00:02:27:0x6327E424 :State change from (STATE_ACTIVE, SUBSTATE_NONE)  to (STATE_ACTIVE,
SUBSTATE_CONNECTING)
00:02:27:0x6327E424 :State change from (STATE_ACTIVE, SUBSTATE_CONNECTING)  to
(STATE_ACTIVE, SUBSTATE_CONNECTING)
00:02:27:udpsock_close_connect:Socket fd:2 closed for connid 2 with remote port:5060
00:02:27:CCSIP-SPI-CONTROL: sipSPICheckSocketConnection:Connid(1) created to
172.18.200.237:5060, local_port 54607
00:02:27:0x6327E424 :State change from (STATE_ACTIVE, SUBSTATE_CONNECTING)  to
(STATE_ACTIVE, SUBSTATE_NONE)
00:02:27:CCSIP-SPI-CONTROL: act_active_connection_created Call Disconnect - Sending Bye
00:02:27:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
00:02:27:sip_stats_method
00:02:27:0x6327E424 :State change from (STATE_ACTIVE, SUBSTATE_NONE)  to
(STATE_DISCONNECTING, SUBSTATE_NONE)
00:02:27:Sent:
BYE sip:2021010124@172.18.200.237:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Mon, 01 Mar 1993 00:02:12 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:1
Timestamp:730944147
CSeq:103 BYE
Content-Length:0

00:02:27:Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:58:08 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Server:Cisco-SIPGateway/IOS-12.x
Timestamp:730944147
Content-Length:0
CSeq:103 BYE

00:02:27:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:172.18.200.237:5060
00:02:27:CCSIP-SPI-CONTROL: act_disconnecting_new_message
00:02:27:CCSIP-SPI-CONTROL: sact_disconnecting_new_message_response
00:02:27:CCSIP-SPI-CONTROL: sipSPICheckResponse
```

```
00:02:27:sip_stats_status_code
00:02:27: Roundtrip delay 16 milliseconds for method BYE

00:02:27:CCSIP-SPI-CONTROL: sipSPICallCleanup
00:02:27:sipSPIIcpifUpdate :CallState:4 Playout:0 DiscTime:14742 ConnTime 13360

00:02:27:0x6327E424 :State change from (STATE_DISCONNECTING, SUBSTATE_NONE)  to
(STATE_DEAD, SUBSTATE_NONE)
00:02:27:The Call Setup Information is :
Call Control Block (CCB) :0x6327E424
State of The Call        :STATE_DEAD
TCP Sockets Used         :NO
Calling Number           :888001
Called Number            :2021010124
Negotiated Codec         :g711ulaw
Negotiated Codec Bytes   :160
Negotiated Dtmf-relay    :0
Dtmf-relay Payload       :0

00:02:27:
 Source IP Address (Sig  ):10.15.66.43
Source IP Address (Media):10.15.66.43
Source IP Port    (Media):16398
Destn  IP Address (Media):172.18.200.237
Destn  IP Port    (Media):16898
Destn SIP Req Addr:Port  :172.18.200.237:5060
Destn SIP Resp Addr:Port :0.0.0.0:0
Destination Name         :172.18.200.237

00:02:27:
 Orig Destn IP Address:Port (Media):0.0.0.0:0

00:02:27:
 Disconnect Cause (CC)    :16
Disconnect Cause (SIP)   :200

00:02:27:****Deleting from UAC table

00:02:27:Removing call id 3

00:02:27:RequestCloseConnection:Closing connid 1 Local Port 54607
00:02:27:Queued event from SIP SPI :SIPSPI_EV_CLOSE_CONNECTION (8)
00:02:27: freeing ccb 6327E424

00:02:27:udpsock_close_connect:Socket fd:1 closed for connid 1 with remote port:5060
Router-3640#

Router-5300# debug ccsip all

All SIP call tracing enabled

Router-5300#
3d04h:Received:
INVITE sip:2021010124@172.18.200.237:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>
Date:Mon, 01 Mar 1993 00:02:12 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Supported:timer,100rel
Min-SE: 1800
Cisco-Guid:3563045876-351146444-2147852364-2382746380
User-Agent:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
```

```
Max-Forwards:1
Timestamp:730944132
Contact:<sip:888001@10.15.66.43:5060;user=phone>
Expires:60
Allow-Events:telephone-event
Content-Type:application/sdp
Content-Length:291

v=0
o=CiscoSystemsSIP-GW-UserAgent 9502 9606 IN IP4 10.15.66.43
s=SIP Call
c=IN IP4 10.15.66.43
t=0 0
m=audio 16398 RTP/AVP 0 100 101
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=direction:active

3d04h:HandleUdp
SocketReads :Msg enqueued for SPI with IPaddr:172.18.200.238:10105
3d04h:CCSIP-SPI-CONTROL: sipSPISipIncomingMsg
3d04h:0x629748AC :State change from (UNDEFINED, SUBSTATE_NONE)  to (STATE_IDLE,
SUBSTATE_NONE)
3d04h:CCSIP-SPI-CONTROL: act_idle_new_message
3d04h:CCSIP-SPI-CONTROL: Clock Time Zone is UTC, same as GMT:Using GMT
3d04h:CCSIP-SPI-CONTROL: sact_idle_new_message_invite
3d04h:sip_stats_method
3d04h:CCSIP-SPI-CONTROL: sipSPIUASSessionTimer
3d04h:sipSPIGetSdpBody :Parse incoming session description
CCSIP-SPI-CONTROL: (4294967295) Warning:No network type specified in comediadir attribute.
3d04h:****Deleting from UAS Request table

3d04h:sipSPIUdeleteCcbFromTable:Entry not found for search key

3d04h:CCSIP-SPI-CONTROL: sipSPIMatchSrcIpGroup
3d04h:CCSIP-SPI-CONTROL: sipSPIContinueNewMsgInvite
3d04h:sipSPIContinueNewMsgInvite:non dial peer leg - using RTP Supported Codecs

3d04h:sipSPIContinueNewMsgInvite:RTP Preferred Codecs supported by GW 18

3d04h:sipSPIContinueNewMsgInvite:RTP Preferred Codecs supported by GW 0

3d04h:sipSPIContinueNewMsgInvite:RTP Preferred Codecs supported by GW 8

3d04h:sipSPIContinueNewMsgInvite:RTP Preferred Codecs supported by GW 4

3d04h:sipSPIContinueNewMsgInvite:RTP Preferred Codecs supported by GW 2

3d04h:sipSPIContinueNewMsgInvite:RTP Preferred Codecs supported by GW 15

3d04h:sipSPIContinueNewMsgInvite:RTP Preferred Codecs supported by GW 3

3d04h:sipSPIDoFaxMediaNegotiation()
3d04h:sipSPIDoMediaNegotiation:Codec (g711ulaw) Negotiation Successful on Static Payload

3d04h: sipSPIDoPtimeNegotiation:One ptime attribute found - value:20
3d04h: convert_ptime_to_codec_bytes:Values :Codec:g711ulaw ptime :20, codecbytes:160

3d04h: convert_codec_bytes_to_ptime:Values :Codec:g711ulaw codecbytes :160, ptime:20
```

```
3d04h: Parsed the SDP for direction:Extraction of src address triggered with role as :1

3d04h:sipSPIDoDTMFRelayNegotiation:Requested DTMF-RELAY option(s) not found in Preferred
DTMF-RELAY option list!
3d04h: sipSPIDoMediaNegotiation:DTMF Relay mode :Inband Voice

3d04h:sip_sdp_get_modem_relay_cap_params:
3d04h:sip_sdp_get_modem_relay_cap_params:NSE payload from X-cap = 0
3d04h:sip_do_nse_negotiation:NSE Payload 100 found in SDP
3d04h:sip_do_nse_negotiation:Remote NSE payload = local one = 100, Use it
3d04h:sip_select_modem_relay_params:X-tmr not present in SDP. Disable modem relay
3d04h:sipSPIUpdCcbWithSdpInfo:SDP Media Information:
Negotiated Codec      :g711ulaw , bytes :160
Early Media           :0
Delayed Media         :0
Bridge Done           :0
New Media             :0
DSP DNLD Reqd         :0
Media Dest addr/Port  :10.15.66.43:16398
Orig Media Addr/Port  :0.0.0.0:0

3d04h:sipSPIHandleInviteMedia:
Negotiated Codec        :g711ulaw, bytes :160
Preferred Codec         :g729r8, bytes :20
Preferred  DTMF relay 1 :0
Preferred  DTMF relay 2 :0
Negotiated DTMF relay   :0
Preferred and Negotiated NTE payloads:101 0
Preferred and Negotiated NSE payloads:100 100
Preferred and Negotiated Modem Relay:0 0
Preferred and Negotiated Modem Relay GwXid:1 0

3d04h:sipSPIContinueNewMsgInvite:Requires reliable-provisional support
3d04h:sipSPIDoQoSNegotiation - SDP body with media description
3d04h:sipSPIAddBillingInfoToCcb:sipCallId for billing records =
D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
3d04h:****Adding to UAS Request table

3d04h:adding call id 31 to table

3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_status_code
3d04h:****Adding to UAS Response table

3d04h:Previous Hop 10.15.66.43:5060

3d04h:0x629748AC :State change from (STATE_IDLE, SUBSTATE_NONE)  to (STATE_RECD_INVITE,
SUBSTATE_NONE)
3d04h:Sent:
SIP/2.0 100 Trying
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:57:53 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Timestamp:730944132
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Allow-Events:telephone-event
Content-Length:0



3d04h:Queued event from SIP SPI :SIPSPI_EV_CC_CALL_PROCEEDING (11)
```

```
3d04h:ccsip_report_digit_control:enable=0:
3d04h: ccsip_report_digit_control:disabled.
3d04h:CCSIP-SPI-CONTROL: act_recdinvite_proceeding
3d04h:Queued event from SIP SPI :SIPSPI_EV_CC_CALL_ALERTING (13)
3d04h:CCSIP-SPI-CONTROL: sipSPIIncomingCallSDP
3d04h:sipSPIUpdateSrcSdpFixedPart
3d04h:sipSPIUpdateSrcSdpVariablePart
3d04h:sipSPIUpdateSrcSdpVariablePart Negotiated NSE payload :100

3d04h:sipSPIRtcpUpdates:rtcp_session info
                laddr = 172.18.200.237, lport = 16898, raddr = 10.15.66.43, rport=16398
3d04h:sipSPIRtcpUpdates:callback- Update the actual remote media source information

3d04h: sipSPIRtcpUpdates No rtp session in bridge, create a new one

3d04h:CCSIP-SPI-CONTROL: ccsip_caps_ind
3d04h:ccsip_get_rtcp_session_parameters:CURRENT VALUES:
ccCallID=49, current_seq_num=0x2059
3d04h:ccsip_get_rtcp_session_parameters:NEW VALUES:
ccCallID=49, current_seq_num=0x944
3d04h:ccsip_caps_ind:Load DSP with negotiated codec :g711ulaw, Bytes=160
3d04h:sipSPISetDTMFRelayMode:set DSP for dtmf-relay =
CC_CAP_DTMF_RELAY_INBAND_VOICE_AND_OOB
3d04h:sip_set_modem_caps:Negotiation already Done. Set negotiated Modem caps
3d04h:sip_set_modem_caps:Modem Relay & Passthru both disabled
3d04h:sip_set_modem_caps:nse payload = 100, ptru mode = 0, ptru-codec=0, redundancy=0,
xid=0, relay=0, sprt-retry=12, latecncy=200, compres-dir=3, dict=1024, strnlen=32
3d04h:ccsip_caps_ind:Load DSP with codec :g711ulaw, Bytes=160
3d04h:CCSIP-SPI-CONTROL: ccsip_caps_ack
3d04h:CCSIP-SPI-CONTROL: act_recdinvite_alerting
3d04h:Session Type is Media/Qos/Security, SDP body is attached
3d04h:CCSIP-SPI-CONTROL: sipSPIIncomingCallSDP
3d04h: SDP already there use old sdp and updatemedia if needed

3d04h:sipSPIUpdateSrcSdpVariablePart
3d04h:sipSPIUpdateSrcSdpVariablePart Negotiated NSE payload :100

3d04h:sipSPIAddLocalContact
3d04h:sip_generate_sdp_xcaps_list:Modem Relay disabled. X-cap not needed

3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_status_code
3d04h:0x629748AC :State change from (STATE_RECD_INVITE, SUBSTATE_NONE)  to
(STATE_RECD_INVITE, SUBSTATE_RECD_INVITE_RECD_PROGRESS)
3d04h:Sent:
SIP/2.0 183 Session Progress
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:57:53 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Timestamp:730944132
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Require:100rel
RSeq:5975
Allow-Events:telephone-event
Contact:<sip:2021010124@172.18.200.237:5060;user=phone>
Content-Type:application/sdp
Content-Disposition:session;handling=required
Content-Length:240

v=0
o=CiscoSystemsSIP-GW-UserAgent 1692 40 IN IP4 172.18.200.237
```

```
s=SIP Call
c=IN IP4 172.18.200.237
t=0 0
m=audio 16898 RTP/AVP 0 100
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=ptime:20
a=direction:passive

3d04h:Queued event from SIP SPI :SIPSPI_EV_CC_CALL_CONNECT (14)
3d04h:CCSIP-SPI-CONTROL: act_sentalert_connect
3d04h:sipSPIUpdCcbWithSdpInfo Bridge was done and there are no fqdn queries in progress,
do RTCP updates

3d04h:sipSPIRtcpUpdates:rtcp_session info
               laddr = 172.18.200.237, lport = 16898, raddr = 10.15.66.43, rport=16398
3d04h:sipSPIRtcpUpdates:callback- Update the actual remote media source information

3d04h: sipSPIRtcpUpdates rtp session already created in bridge - update

3d04h:sipSPIUpdCcbWithSdpInfo:SDP Media Information:
Negotiated Codec      :g711ulaw , bytes :160
Early Media           :2
Delayed Media         :0
Bridge Done           :1048576
New Media             :0
DSP DNLD Reqd         :0
Media Dest addr/Port  :10.15.66.43:16398
Orig Media Addr/Port  :0.0.0.0:0

3d04h:sipSPIProcessMediaChanges
3d04h:CCSIP-SPI-CONTROL: sipSPIIncomingCallSDP
3d04h: SDP already there use old sdp and updatemedia if needed

3d04h:sipSPIUpdateSrcSdpVariablePart
3d04h:sipSPIUpdateSrcSdpVariablePart Negotiated NSE payload :100

3d04h:sipSPIAddLocalContact
3d04h:sip_generate_sdp_xcaps_list:Modem Relay disabled. X-cap not needed

3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_status_code
3d04h:0x629748AC :State change from (STATE_RECD_INVITE,
SUBSTATE_RECD_INVITE_RECD_PROGRESS)  to (STATE_SENT_SUCCESS, SUBSTATE_NONE)
3d04h:Sent:
SIP/2.0 200 OK
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:57:53 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Timestamp:730944132
Server:Cisco-SIPGateway/IOS-12.x
CSeq:101 INVITE
Allow-Events:telephone-event
Contact:<sip:2021010124@172.18.200.237:5060;user=phone>
Content-Type:application/sdp
Content-Length:240

v=0
o=CiscoSystemsSIP-GW-UserAgent 1692 40 IN IP4 172.18.200.237
s=SIP Call
c=IN IP4 172.18.200.237
```

```
t=0 0
m=audio 16898 RTP/AVP 0 100
a=rtpmap:0 PCMU/8000
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194
a=ptime:20
a=direction:passive

3d04h:Received:
PRACK sip:2021010124@172.18.200.237:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Mon, 01 Mar 1993 00:02:12 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
CSeq:102 PRACK
RAck:5975 101 INVITE
Content-Length:0

3d04h:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:172.18.200.238:50689
3d04h:*****CCB found in UAS Request table

3d04h:CCSIP-SPI-CONTROL: act_sentsucc_new_message
3d04h:CCSIP-SPI-CONTROL: Clock Time Zone is UTC, same as GMT:Using GMT
3d04h:sip_stats_method
3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_status_code
3d04h:Sent:
SIP/2.0 200 OK
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:57:54 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Server:Cisco-SIPGateway/IOS-12.x
CSeq:102 PRACK
Content-Length:0

3d04h: sipSPIUpdateMediaSrcInfo RTP session to be updated with the new src info OLD
addr:port 10.15.66.43:16398 , NEW addr:port 172.18.200.238:16398

3d04h:Received:
ACK sip:2021010124@172.18.200.237:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Mon, 01 Mar 1993 00:02:12 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Max-Forwards:1
Content-Length:0
CSeq:101 ACK

3d04h:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:172.18.200.238:50689
3d04h:*****CCB found in UAS Request table

3d04h:CCSIP-SPI-CONTROL: act_sentsucc_new_message
3d04h:CCSIP-SPI-CONTROL: Clock Time Zone is UTC, same as GMT:Using GMT
3d04h:sip_stats_method
3d04h:0x629748AC :State change from (STATE_SENT_SUCCESS, SUBSTATE_NONE)  to (STATE_ACTIVE,
SUBSTATE_NONE)
3d04h:The Call Setup Information is :
Call Control Block (CCB) :0x629748AC
State of The Call        :STATE_ACTIVE
TCP Sockets Used         :NO
```

```
Calling Number           :888001
Called Number            :2021010124
Negotiated Codec         :g711ulaw
Negotiated Codec Bytes   :160
Negotiated Dtmf-relay    :0
Dtmf-relay Payload       :0

3d04h:
 Source IP Address (Sig  ):172.18.200.237
Source IP Address (Media):172.18.200.237
Source IP Port    (Media):16898
Destn  IP Address (Media):172.18.200.238
Destn  IP Port    (Media):16398
Destn SIP Req Addr:Port  :10.15.66.43:5060
Destn SIP Resp Addr:Port :172.18.200.238:5060
Destination Name         :172.18.200.238

3d04h:
 Orig Destn IP Address:Port (Media):10.15.66.43:16398

3d04h:%ISDN-6-CONNECT:Interface Serial0:29 is now connected to 2021010124
3d04h:%ISDN-6-DISCONNECT:Interface Serial0:29  disconnected from 2021010124 , call lasted
13 seconds
3d04h:Queued event from SIP SPI :SIPSPI_EV_CC_CALL_DISCONNECT (15)
3d04h:CCSIP-SPI-CONTROL: act_active_disconnect
3d04h:Queued event from SIP SPI :SIPSPI_EV_CREATE_CONNECTION (6)
3d04h:0x629748AC :State change from (STATE_ACTIVE, SUBSTATE_NONE)  to (STATE_ACTIVE,
SUBSTATE_CONNECTING)
3d04h:0x629748AC :State change from (STATE_ACTIVE, SUBSTATE_CONNECTING)  to (STATE_ACTIVE,
SUBSTATE_CONNECTING)
3d04h:CCSIP-SPI-CONTROL: sipSPICheckSocketConnection:Connid(1) created to
10.15.66.43:5060, local_port 53993
3d04h:0x629748AC :State change from (STATE_ACTIVE, SUBSTATE_CONNECTING)  to (STATE_ACTIVE,
SUBSTATE_NONE)
3d04h:CCSIP-SPI-CONTROL: act_active_connection_created Call Disconnect - Sending Bye
3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_method
3d04h:0x629748AC :State change from (STATE_ACTIVE, SUBSTATE_NONE)  to
(STATE_DISCONNECTING, SUBSTATE_NONE)
3d04h:Sent:
BYE sip:888001@10.15.66.43:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  172.18.200.237:5060
From:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
To:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
Date:Tue, 04 Jan 2000 23:57:54 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:1
Timestamp:947030287
CSeq:101 BYE
Content-Length:0

3d04h:CCSIP-SPI-CONTROL: act_disconnecting_wait_200
3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_method
3d04h:Sent:
BYE sip:888001@10.15.66.43:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  172.18.200.237:5060
From:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
To:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
Date:Tue, 04 Jan 2000 23:57:54 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:1
```

```
Timestamp:947030287
CSeq:101 BYE
Content-Length:0

3d04h:CCSIP-SPI-CONTROL: act_disconnecting_wait_200
3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_method
3d04h:Sent:
BYE sip:888001@10.15.66.43:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  172.18.200.237:5060
From:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
To:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
Date:Tue, 04 Jan 2000 23:57:54 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:1
Timestamp:947030287
CSeq:101 BYE
Content-Length:0

3d04h:CCSIP-SPI-CONTROL: act_disconnecting_wait_200
3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_method
3d04h:Sent:
BYE sip:888001@10.15.66.43:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  172.18.200.237:5060
From:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
To:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
Date:Tue, 04 Jan 2000 23:57:54 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:1
Timestamp:947030288
CSeq:101 BYE
Content-Length:0

3d04h:Received:
BYE sip:2021010124@172.18.200.237:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Mon, 01 Mar 1993 00:02:12 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:1
Timestamp:730944147
CSeq:103 BYE
Content-Length:0

3d04h:HandleUdpSocketReads :Msg enqueued for SPI with IPaddr:172.18.200.238:54607
3d04h:*****CCB found in UAS Request table

3d04h:CCSIP-SPI-CONTROL: act_disconnecting_new_message
3d04h:CCSIP-SPI-CONTROL: sact_disconnecting_new_message_request
3d04h:CCSIP-SPI-CONTROL: Clock Time Zone is UTC, same as GMT:Using GMT
3d04h:sip_stats_method
3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_status_code
3d04h:Sent:
SIP/2.0 200 OK
Via:SIP/2.0/UDP  10.15.66.43:5060
From:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
To:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
Date:Tue, 04 Jan 2000 23:58:08 GMT
```

```
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
Server:Cisco-SIPGateway/IOS-12.x
Timestamp:730944147
Content-Length:0
CSeq:103 BYE

3d04h:CCSIP-SPI-CONTROL: act_disconnecting_wait_200
3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_method
3d04h:Sent:
BYE sip:888001@10.15.66.43:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  172.18.200.237:5060
From:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
To:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
Date:Tue, 04 Jan 2000 23:58:08 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:1
Timestamp:947030288
CSeq:101 BYE
Content-Length:0

3d04h:CCSIP-SPI-CONTROL: act_disconnecting_wait_200
3d04h:Queued event from SIP SPI :SIPSPI_EV_SEND_MESSAGE (7)
3d04h:sip_stats_method
3d04h:Sent:
BYE sip:888001@10.15.66.43:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP  172.18.200.237:5060
From:<sip:2021010124@172.18.200.237;user=phone>;tag=1069B954-25F
To:"888001" <sip:888001@10.15.66.43>;tag=20694-C53
Date:Tue, 04 Jan 2000 23:58:08 GMT
Call-ID:D6EB9E87-14EE11CC-8008A04C-8E05D30C@10.15.66.43
User-Agent:Cisco-SIPGateway/IOS-12.x
Max-Forwards:1
Timestamp:947030290
CSeq:101 BYE
Content-Length:0

3d04h:CCSIP-SPI-CONTROL: act_disconnecting_wait_200
3d04h:CCSIP-SPI-CONTROL: act_disconnecting_wait_200 :Out of retries
3d04h:CCSIP-SPI-CONTROL: sipSPICallCleanup
3d04h:sipSPIIcpifUpdate :CallState:4 Playout:0 DiscTime:27538424 ConnTime 27536491

3d04h:0x629748AC :State change from (STATE_DISCONNECTING, SUBSTATE_NONE)  to (STATE_DEAD,
SUBSTATE_NONE)
3d04h:The Call Setup Information is :
Call Control Block (CCB) :0x629748AC
State of The Call        :STATE_DEAD
TCP Sockets Used         :NO
Calling Number           :888001
Called Number            :2021010124
Negotiated Codec         :g711ulaw
Negotiated Codec Bytes   :160
Negotiated Dtmf-relay    :0
Dtmf-relay Payload       :0

3d04h:
 Source IP Address (Sig  ):172.18.200.237
Source IP Address (Media):172.18.200.237
Source IP Port    (Media):16898
Destn  IP Address (Media):172.18.200.238
Destn  IP Port    (Media):16398
Destn SIP Req Addr:Port  :10.15.66.43:5060
Destn SIP Resp Addr:Port :172.18.200.238:5060
```

```
Destination Name         :172.18.200.238

3d04h:
 Orig Destn IP Address:Port (Media):10.15.66.43:16398

3d04h:
 Disconnect Cause (CC)    :16
Disconnect Cause (SIP)   :500

3d04h:****Deleting from UAS Request table

3d04h:****Deleting from UAS Response table

3d04h:Removing call id 31

3d04h:RequestCloseConnection:Closing connid 1 Local Port 53993
3d04h:Queued event from SIP SPI :SIPSPI_EV_CLOSE_CONNECTION (8)
3d04h: freeing ccb 629748AC

3d04h:udpsock_close_connect:Socket fd:1 closed for connid 1 with remote port:5060NAT-5300#
Router-5300#
```

Additional References

For additional information related to Connection-Oriented Media (Comedia) Enhancements for SIP, refer to the following references:

The following sections provide additional references related to this feature:

Related Documents

Related Topic	Document Title
SIP configuration tasks	*Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2, "Configuring Session Initiation Protocol for Voice over IP" chapter
IP configuration tasks	*Cisco IOS IP Configuration Guide*, Release 12.2
IP configuration commands	*Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2
VoIP configuration tasks	*Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
NAT configuration tasks	*Configuring Network Address Translation: Getting Started*

Standards

Standards[1]	Title
draft-ietf-mmusic-sdp-comedia-04.txt	*Connection-Oriented Media Transport in SDP*

1. Not all supported standards are listed.

MIBs

MIBs[1]	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 obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB website on Cisco.com at the following URL: http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

1. Not all supported MIBs are listed.

To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:

http://tools.cisco.com/ITDIT/MIBS/servlet/index

If Cisco MIB Locator does not support the MIB information that you need, you can also obtain a list of supported MIBs and download MIBs from the Cisco MIBs page at the following URL:

http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

To access Cisco MIB Locator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions found at this URL:

http://www.cisco.com/register

RFCs

RFCs[1]	Title
RFC 2543	*SIP: Session Initiation Protocol*

1. Not all supported RFCs are listed.

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, tools, and lots more. 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 the new command that configures Connection-Oriented Media (Comedia) Enhancements for SIP feature. All other commands used with this feature are documented in the Cisco IOS Release 12.2T command reference publications.

- **nat symmetric check media-src**
- **nat symmetric role**

nat symmetric check media-src

To enable the gateway to check the media source of incoming Real-time Transport Protocol (RTP) packets in symmetric Network Address Translation (NAT) environments, use the **nat symmetric check media-src** command in SIP user agent configuration mode. To disable media source checking, use the **no** form of this command.

nat symmetric check media-src

no nat symmetric check media-src

Syntax Description

This command has no arguments or keywords.

Defaults

Default behavior is media source checking enabled.

Command Modes

SIP user agent configuration

Command History

Release	Modification
12.2(13)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850, and Cisco CVA 120 series.

Usage Guidelines

This command provides the ability to enable or disable symmetric NAT settings for the Session Initiation Protocol (SIP) user agent. Use the **nat symmetric check media-src** command to configure the gateway to check the media source address and port of the first incoming Realtime Transport Protocol (RTP) packet. Use the **nat symmetric role** command to set the symmetric NAT endpoint role to **active** or **passive**.

Examples:

The following example enables checking the media source:

```
router (config)# sip-ua
router (config-sip-ua)# nat symmetric check-media-src
```

Related Commands

Command	Description
debug ccsip all	Enables all SIP-related debugging.
debug ip nat sip	Displays information about SIP messages and IP packets translated by the IP NAT feature
nat symmetric role	Specifies NAT endpoint settings

nat symmetric role

To define endpoint settings to initiate or accept a connection for symmetric Network Address Translation (NAT) configuration, use the **nat symmetric role** command in SIP user agent configuration mode. To disable **nat symmetric role**, use the **no** form of this command.

nat symmetric role {**active** | **passive**}

no nat symmetric role {active | passive}

Syntax Description

role {active | passive}	(Optional) Sets the symmetric NAT endpoint role to active, originating an outgoing connection, or passive, accepting an incoming connection to the port number on the m= line of the Session Description Protocol (SDP) body sent from itself to the other endpoint.

Defaults

Default behavior is the NAT direction role set to passive.

Command Modes

SIP user agent configuration

Command History

Release	Modification
12.2(13)T	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850, and Cisco CVA 120 series.

Usage Guidelines

This command provides the ability to specify symmetric NAT endpoint settings for the Session Initiation Protocol (SIP) user agent. Use the **nat symmetric check media-src** command to configure the gateway to check the media source address and port of the first incoming Realtime Transport Protocol (RTP) packet. Checking for media packets is automatically enabled if the gateway receives the direction role active or both. Otherwise, use the **nat symmetric role** command to set the symmetric NAT endpoint role to **active** or **passive**. We recommend that you use the **nat symmetric role** command under the following conditions:

- Endpoints are aware of their presence inside or outside of NAT
- Endpoints parse and process direction:<role> in Session Description Protocol (SDP)

Otherwise configuring the **nat symmetric role** command may not achieve the desired results.

Examples:

The following example enables checking the media source:

```
router (config)# sip-ua
router (config-sip-ua)# nat symmetric check-media-src
```

The following example sets the endpoint role in connection setup to active:

```
router (config)# sip-ua
router (config-sip-ua)# nat symmetric role active
```

Related Commands

Command	Description
debug ccsip all	Enables all SIP-related debugging.
debug ip nat sip	Displays information about SIP messages and IP packets translated by the IP NAT feature
nat symmetric check media-src	Enables source media checking for symmetric NAT

Glossary

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

call—In SIP, a call consists of all participants in a conference that 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.

call-ID—A general header that uniquely identifies a particular invitation or all registrations of a particular client.

CLI—command-line interface.

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 insight regarding the capabilities of the called and calling parties.

IP—Internet protocol. A connectionless protocol that operates at the network layer (Layer 3) of the OSI model. IP provides features for addressing, type-of-service specification, fragmentation and reassemble, and security. Defined in RFC 791. This protocol works with TCP and is usually identified as TCP/IP. See TCP/IP.

originator—User agent that initiates the transfer or Refer request with the recipient.

RTCP—RTP Control Protocol. The protocol monitors an RTP connection and conveys information about the ongoing session.

RTP—Real-Time Transport Protocol. The protocol provides end-to-end network transport functions for applications sending real-time data and services such as payload type identification, sequence numbering, time-stamping, and delivery monitoring. A network protocol used to carry packetized audio and video traffic over an IP network.

SDP—Session Description Protocol. Messages containing capabilities information that are exchanged between gateways.

session—A SIP session includes a set of multimedia senders and receivers and the data streams flowing 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.

SIP URL—Session Initiation Protocol Uniform Resource Locator. Used in SIP messages to indicate the originator, recipient, and destination of the SIP request. Takes the basic form of *user@host*, where *user* is a name or telephone number, and *host* is a domain name or network address.

SPI—Service Provider Interface. A general category for VoIP protocols.

TCP—Transmission Control Protocol. Connection-oriented transport layer protocol that provides reliable full-duplex data transmissions. TCP is part of the TCP/IP protocol stack. See also TCP/IP and IP.

TCP/IP—Transmission Control Protocol/Internet Protocol. Common name for the suite of protocols developed by the U.S. Department of Defense in the 197's to support the construction of worldwide internetworks. TCP and IP are the two best known protocols in the suite. See also TCP and IP.

TEL URL—Telephone Uniform Resource Locator. Describes voice call connections to a terminal. Can also be any connection through a voice messaging system or a service that can be operated using DTMF tones. Takes the basic form of *tel:telephone subscriber number*, where *tel* indicates a URL and requests the local entity to place a voice call, and *telephone subscriber number* is the number to receive the call.

UA—user agent. A combination of UAS and UAC that initiates and receives calls. See UAS and UAC.

UAC—user agent client. A client application that initiates a SIP request.

UAS—user agent server. A server application that contacts the user when a SIP request is received, then returns a response on behalf of the user. The response accepts, rejects, or redirects the request.

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

URL—Uniform Resource Locator. Standard address of any resource on the Internet.

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