Troubleshoot a SIP Call Between Two Endpoints

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Contents

Introduction

This document provides a sample configuration of two fax machines in order to demonstrate how a Session Initiation Protocol (SIP) call takes place between two gateways. This document also provides an explanation on the output of the **debug ccsip messages** command for troubleshooting SIP call failures.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

The information in this document is based on these software and hardware versions:

- Two fax machines
- VG224 that runs Cisco IOS® Software Release 12.4(4)T1
- Cisco 3745 router that runs Cisco IOS Software Release 12.3(11)T8

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Conventions

Refer to Cisco Technical Tips Conventions for more information on document conventions.

Configure

In this section, you are presented with the information to configure the features described in this document.

Note: Use the Command Lookup Tool (registered customers only) to find more information on the commands used in this document.

Network Diagram

This document uses this network setup:

Configurations

This document uses these configurations:

- VG224
- Cisco 3745

VG224

```
vg224#show run
Building configuration...
!
voice call send-alert
voice rtp send-recv
!
voice service pots
!
voice service voip
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco
 sip
  bind control source-interface FastEthernet0/0
  bind media source-interface FastEthernet0/0
!
voice-port 2/0
 idle-voltage low
!
dial-peer voice 1 pots
<fax machine connected to this port>
 destination-pattern 9000
 port 2/0
!
dial-peer voice 100 voip
 destination-pattern 8000
 no modem passthrough
 session protocol sipv2
 session target ipv4:172.16.184.83
 incoming called-number .
 codec g711ulaw
```

```
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco
!
```

Cisoc 3745

```
HTTS-VRK1-3745-1#show run
Building configuration...
!
voice service voip
 sip
  bind control source-interface FastEthernet0/0
  bind media source-interface FastEthernet0/0
!
!
voice-port 4/1/0
!
!
dial-peer voice 9000 voip
 destination-pattern 9000
 session protocol sipv2
 session target ipv4:172.16.13.87
 incoming called-number .
 codec g711ulaw
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco
 no vad
!
dial-peer voice 9 pots
 destination-pattern 8000
 fax rate voice
 port 4/1/0
 forward-digits all
```

Verify

There is currently no verification procedure available for this configuration.

Troubleshoot

Use this section to troubleshoot your configuration.

The Output Interpreter Tool (registered customers only) (OIT) supports certain **show** commands. Use the OIT to view an analysis of **show** command output.

Note: Refer to Important Information on Debug Commands before you use **debug** commands.

This is the output of the **debug ccsip messages** command:

```
!--- This is the first invite message sent out
!--- to the terminating SIP gateway.
!--- This is similar to a setup message in H.323 or Q.931.

*Mar 1 00:33:42.419: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:8000@172.16.184.83:5060 SIP/2.0

!--- 8000 is the DN of the call, 172.16.184.83 is
!--- the IP address of the remote gateway, and
!--- 5060 is the port the SIP works on.
!--- This configuration uses SIP version 2.0.
```

```
Via: SIP/2.0/UDP 172.16.13.87:5060;branch=z9hG4bKB21AF

!--- The VIA field is used for devices in the patch that
!--- need to be aware of the call.
!--- In this case, there are no SIP devices in between the two gateways.

Remote-Party-ID: <sip:9000@172.16.13.87>;party=calling;screen=no;privacy=off

!--- The DN and URI of the remote SIP device that is called.

From: <sip:9000@172.16.13.87>;tag=1EDC10-2436
To: <sip:8000@172.16.184.83>
Date: Fri, 01 Mar 2002 00:33:42 GMT

!--- The time that the invite is sent out

Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87

!--- The call ID is unique for every call.
!--- This ID is used to identify a particular call
!--- in a busy router.

Supported: 100rel,timer,resource-priority,replaces
Min-SE: 1800
Cisco-Guid: 3481906499-736235990-2149183265-3714191467
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER,
SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE

!--- The sequence number for each transaction.

Max-Forwards: 70
Timestamp: 1014942822
Contact: <sip:9000@172.16.13.87:5060>

!--- This is the address used to reach the calling party on the return path.

Expires: 180

!--- This message expires in 180 seconds.

Allow-Events: telephone-event
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 215

v=0

!--- The Session Descriptor Protocol (SDP) version is zero.
!--- This is different from the SIP version used
!--- in this example configuration.

o=CiscoSystemsSIP-GW-UserAgent 1715 2724 IN IP4 172.16.13.87

!--- The owner of the device that created the call.
!--- This is sometimes referred to as organization.

s=SIP Call

!--- The name given to this particular SIP call. This is the description.

c=IN IP4 172.16.13.87

!--- Connection information. Usually includes the IP address of
```

```
!--- the originating device. It is an optional field.

t=0 0
m=audio 18080 RTP/AVP 0 19

!--- This is the media information. In this case,
!--- 18080 is used as the UDP port for RTP.

c=IN IP4 172.16.13.87
a=rtpmap:0 PCMU/8000

!--- This is the media attributes. Notice the 0 and 19 in
!--- the media field. These are the
!--- attributes that go with that. PCMU/8000 is G711ulaw.

a=rtpmap:19 CN/8000
a=ptime:20

!--- A packetization period of 20 ms.



!--- In this output, invite, SDP is not a required parameter.
!--- But in this case you see that SDP sent out.
!--- SDP carries information about capabilities.
!--- No information about fax capabilities are
!--- exchanged in the beginning because it is only a voice
!--- call until you hear fax tones from the terminating fax machine.


*Mar 1 00:33:43.203: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.16.13.87:5060;branch=z9hG4bKB21AF
From: <sip:9000@172.16.13.87>;tag=1EDC10-2436
To: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
Date: Tue, 28 Feb 2006 23:43:36 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Timestamp: 1014942822
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Allow-Events: telephone-event
Content-Length: 0


!--- The terminating machine sets up an analog
!--- connection to the fax machine, and while it waits,
!--- it sends a "trying" message. This stops the
!--- originating gateway from sending another invite.


*Mar 1 00:33:43.207: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.16.13.87:5060;branch=z9hG4bKB21AF
From: <sip:9000@172.16.13.87>;tag=1EDC10-2436
To: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
Date: Tue, 28 Feb 2006 23:43:36 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Timestamp: 1014942822
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Require: 100rel
RSeq: 3696
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE,
NOTIFY, INFO, UPDATE, REGISTER
```

```
Allow-Events: telephone-event
Contact: <sip:8000@172.16.184.83:5060>
Content-Disposition: session;handling=required
Content-Type: application/sdp
Content-Length: 194

v=0
o=CiscoSystemsSIP-GW-UserAgent 7643 2735 IN IP4 172.16.184.83
s=SIP Call
c=IN IP4 172.16.184.83
t=0 0
m=audio 18304 RTP/AVP 0

!--- This is a different UDP port for the reverse direction.

c=IN IP4 172.16.184.83
a=rtpmap:0 PCMU/8000
a=ptime:20


!--- A "progress" indicator tells you that the remote gateway sent a connect
!--- and the fax machine is ringing at this time.
!--- Note that the To and From headers do not change despite
!--- the fact that the message comes in the opposite direction.


*Mar 1 00:33:43.211: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.16.13.87:5060;branch=z9hG4bKB21AF
From: <sip:9000@172.16.13.87>;tag=1EDC10-2436
To: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
Date: Tue, 28 Feb 2006 23:43:36 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Timestamp: 1014942822
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Require: 100rel
RSeq: 3696
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE,
NOTIFY, INFO, UPDATE, REGISTER
Allow-Events: telephone-event
Contact: <sip:8000@172.16.184.83:5060>
Content-Disposition: session;handling=required
Content-Type: application/sdp
Content-Length: 194

v=0
o=CiscoSystemsSIP-GW-UserAgent 7643 2735 IN IP4 172.16.184.83
s=SIP Call
c=IN IP4 172.16.184.83
t=0 0
m=audio 18304 RTP/AVP 0
c=IN IP4 172.16.184.83
a=rtpmap:0 PCMU/8000
a=ptime:20


!--- A provisional ack to the progress message.


*Mar 1 00:33:43.251: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
PRACK sip:8000@172.16.184.83:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.13.87:5060;branch=z9hG4bKC384
From: <sip:9000@172.16.13.87>;tag=1EDC10-2436
```

```
To: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
Date: Fri, 01 Mar 2002 00:33:42 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
CSeq: 102 PRACK
RAck: 3696 101 INVITE
Max-Forwards: 70
Content-Length: 0
```

!--- This is an OK for the PRACK. You can tell this from the Cseq header.

```
*Mar 1 00:33:44.031: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.16.13.87:5060;branch=z9hG4bKC384
From: <sip:9000@172.16.13.87>;tag=1EDC10-2436
To: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
Date: Tue, 28 Feb 2006 23:43:37 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 102 PRACK
Content-Length: 0
```

!--- An OK is received, which is mandatory for an invite.
!--- The OK has information on the accepted media parameters in the SDP.

```
*Mar 1 00:33:49.431: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.16.13.87:5060;branch=z9hG4bKB21AF
From: <sip:9000@172.16.13.87>;tag=1EDC10-2436
To: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
Date: Tue, 28 Feb 2006 23:43:37 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Timestamp: 1014942822
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE,
NOTIFY, INFO, UPDATE, REGISTER
Allow-Events: telephone-event
Contact: <sip:8000@172.16.184.83:5060>
Content-Type: application/sdp
Content-Length: 194

v=0
o=CiscoSystemsSIP-GW-UserAgent 7643 2735 IN IP4 172.16.184.83
s=SIP Call
c=IN IP4 172.16.184.83
t=0 0
m=audio 18304 RTP/AVP 0
c=IN IP4 172.16.184.83
a=rtpmap:0 PCMU/8000
a=ptime:20
```

!--- The ack for the OK.

```
*Mar 1 00:33:49.443: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
ACK sip:8000@172.16.184.83:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.13.87:5060;branch=z9hG4bKD1A5C
From: <sip:9000@172.16.13.87>;tag=1EDC10-2436
```

```
To: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
Date: Fri, 01 Mar 2002 00:33:42 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Max-Forwards: 70
CSeq: 101 ACK
Content-Length: 0


!--- At this point, the terminating gateway hears fax tones and determines it
!--- has to switch the codec to a
!--- fax codec and sends a re-invite. The re-invite contains
!--- information about the new media
!--- parameters that the terminating gateway wants to change to.


*Mar 1 00:33:55.247: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
INVITE sip:9000@172.16.13.87:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.184.83:5060;branch=z9hG4bK1A735
From: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
To: <sip:9000@172.16.13.87>;tag=1EDC10-2436
Date: Tue, 28 Feb 2006 23:43:49 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Supported: 100rel,timer
Min-SE: 1800
Cisco-Guid: 3481906499-736235990-2149183265-3714191467
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE,
NOTIFY, INFO, UPDATE, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1141170229
Contact: <sip:8000@172.16.184.83:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 399

v=0
o=CiscoSystemsSIP-GW-UserAgent 7643 2736 IN IP4 172.16.184.83
s=SIP Call
c=IN IP4 172.16.184.83
t=0 0
m=image 18304 udptl t38
c=IN IP4 172.16.184.83
a=T38FaxVersion:0
a=T38MaxBitRate:14400

!--- The maximum bit rate that is supported by the terminating gateway.

a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:72
a=T38FaxUdpEC:t38UDPRedundancy

!--- UDP redundancy is enabled.



!--- A trying message is sent and an
!--- attempt is made to determine if T.38 fax-relay is supported.
```

```
*Mar 1 00:33:55.275: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.16.184.83:5060;branch=z9hG4bK1A735
From: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
To: <sip:9000@172.16.13.87>;tag=1EDC10-2436
Date: Fri, 01 Mar 2002 00:33:55 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Allow-Events: telephone-event
Remote-Party-ID: <sip:9000@172.16.13.87>;party=called;screen=no;privacy=off
Content-Length: 0


!--- The OK to the re-invite that specifies that you can
!--- do T.38 fax-relay. The same UDP port is retained.


*Mar 1 00:33:55.275: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.16.184.83:5060;branch=z9hG4bK1A735
From: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
To: <sip:9000@172.16.13.87>;tag=1EDC10-2436
Date: Fri, 01 Mar 2002 00:33:55 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE,
NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Remote-Party-ID: <sip:9000@172.16.13.87>;party=called;screen=no;privacy=off
Contact: <sip:9000@172.16.13.87:5060>
Content-Type: application/sdp
Content-Length: 157

v=0
o=CiscoSystemsSIP-GW-UserAgent 1715 2725 IN IP4 172.16.13.87
s=SIP Call
c=IN IP4 172.16.13.87
t=0 0
m=image 18080 udptl t38
c=IN IP4 172.16.13.87


!--- The ack to the OK is received. At this point, fax transmission occurs.


*Mar 1 00:33:55.719: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
ACK sip:9000@172.16.13.87:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.184.83:5060;branch=z9hG4bK1B21D0
From: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
To: <sip:9000@172.16.13.87>;tag=1EDC10-2436
Date: Tue, 28 Feb 2006 23:43:49 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Max-Forwards: 70
CSeq: 101 ACK
Content-Length: 0


!--- Once the fax transmission is completed,
!--- the BYE is received. The BYE is similar to a
!--- release message in Q.931.
```

```
*Mar 1 00:34:45.515: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
BYE sip:9000@172.16.13.87:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.184.83:5060;branch=z9hG4bK1E1E51
From: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
To: <sip:9000@172.16.13.87>;tag=1EDC10-2436
Date: Tue, 28 Feb 2006 23:44:38 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Timestamp: 1141170279
CSeq: 103 BYE
Reason: Q.850;cause=16

!--- Cause code 16 is a normal disconnect cause.

Content-Length: 0

!--- There should be an OK to every message.

*Mar 1 00:34:45.535: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.16.184.83:5060;branch=z9hG4bK1E1E51
From: <sip:8000@172.16.184.83>;tag=85E9C7C8-A4C
To: <sip:9000@172.16.13.87>;tag=1EDC10-2436
Date: Fri, 01 Mar 2002 00:34:45 GMT
Call-ID: D110EA36-2BE211D6-801CEF21-DD62106B@172.16.13.87
Server: Cisco-SIPGateway/IOS-12.x
Timestamp: 1141170279
CSeq: 103 BYE
Reason: Q.850;cause=16
Content-Length: 0


More information about the attributes:

Session description
v= (protocol version)
o= (owner/creator and session identifier).
s= (session name)
i=* (session information)
u=* (URI of description)
e=* (email address)
p=* (phone number)
c=* (connection information - not required if included in all media)
b=* (bandwidth information)
z=* (time zone adjustments)
k=* (encryption key)
a=* (zero or more session attribute lines)

Time description
t= (time the session is active)
r=* (zero or more repeat times)

Media description
m= (media name and transport address)
i=* (media title)
c=* (connection information - optional if included at session-level)
b=* (bandwidth information)
k=* (encryption key)
a=* (zero or more media attribute lines)
* indicated optional item.

Basic Requests
```

```
INVITE: request from a UAC to initiate a session
ACK: confirms receipt of a final response to INVITE
BYE: sent by either side to end a session
CANCEL: sent to end a call not yet connected
UPDATE: Updates offer for not−yet−established sessions.
REGISTER: UA registers with Registrar Server
NOTIFY: notifies that an event has occurred
REFER: the mechanism to initiate a session transfer
INFO: a means of carrying ?data? in a message body


SIP responses:

1xx: Provisional ? request received, continuing to process the request
2xx: Success − action was successfully received, understood, and accepted
3xx: Redirection − further action needs to be taken in order to complete the request
4xx: Client Error − the request contains bad syntax or cannot be
fulfilled at this server
5xx: Server Error − the server failed to fulfill an apparently valid request
6xx: Global Failure − the request cannot be fulfilled at any server
```

Related Information

- **SDP RFC 2327**
- **SIP RFC 3261**
- **Voice Technology Support**
- **Voice and Unified Communications Product Support**
- **Troubleshooting Cisco IP Telephony**
- **Technical Support & Documentation − Cisco Systems**

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