

두 엔드포인트 간의 SIP 통화 문제 해결

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소개

이 문서에서는 두 게이트웨이 간에 SIP(Session Initiation Protocol) 통화가 발생하는 방법을 보여 주기 위해 두 팩스 장치의 샘플 컨피그레이션을 제공합니다. 이 문서에서는 SIP 통화 실패 문제 해결을 위한 `debug ccsip messages` 명령의 출력에 대한 설명을 제공합니다.

사전 요구 사항

요구 사항

이 문서에 대한 특정 요건이 없습니다.

사용되는 구성 요소

이 문서의 정보는 다음 소프트웨어 및 하드웨어 버전을 기반으로 합니다.

- 팩스 2대
- Cisco IOS® Software 릴리스 12.4(4)T1을 실행하는 VG224
- Cisco IOS Software 릴리스 12.3(11)T8을 실행하는 Cisco 3745 라우터

이 문서의 정보는 특정 랩 환경의 디바이스를 토대로 작성되었습니다. 이 문서에 사용된 모든 디바이스는 초기화된(기본) 컨피그레이션으로 시작되었습니다. 현재 네트워크가 작동 중인 경우, 모든 명령어의 잠재적인 영향을 미리 숙지하시기 바랍니다.

표기 규칙

문서 규칙에 대한 자세한 내용은 [Cisco 기술 팁 표기 규칙을 참고하십시오.](#)

구성

이 섹션에는 이 문서에서 설명하는 기능을 구성하기 위한 정보가 표시됩니다.

참고: [명령 조회 도구](#) ([등록된](#) 고객만 해당)를 사용하여 이 문서에 사용된 명령에 대한 자세한 내용을 확인하십시오.

네트워크 다이어그램

이 문서에서는 다음 네트워크 설정을 사용합니다.



구성

이 문서에서는 다음 구성을 사용합니다.

- [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
!
```

Cisco 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
```

다음을 확인합니다.

현재 이 구성에 대해 사용 가능한 확인 절차가 없습니다.

문제 해결

이 섹션에서는 컨피그레이션 문제를 해결할 수 있습니다.

Output [Interpreter 도구](#)([등록된](#) 고객만 해당)(OIT)는 특정 **show** 명령을 지원합니다. OIT를 사용하여 **show** 명령 출력의 분석을 봅니다.

참고: debug 명령을 사용하기 전에 디버그 [명령에 대한 중요 정보](#)를 참조하십시오.

다음은 debug ccsip messages 명령의 출력입니다.

```
!--- 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/xxxxxxxxxxxxx/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/xxxxxxxxxxxxx/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/xxxxxxxxxxxxx/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/xxxxxxxxxxxxx/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

[관련 정보](#)

- [SDP RFC 2327](#)
- [SIP RFC 3261](#)
- [음성 기술 지원](#)
- [음성 및 통합 커뮤니케이션 제품 지원](#)
- [Cisco IP 텔레포니 문제 해결](#)
- [기술 지원 및 문서 - Cisco Systems](#)