對兩個終端之間的SIP呼叫進行故障排除

目錄

簡介

本文提供兩台傳真機的範例組態，以說明作業階段啟始通訊協定(SIP)通話如何在兩個閘道之間進行。本檔案也提供有關**debug ccsip messages**命令輸出的說明，以疑難排解SIP呼叫失敗。

必要條件

需求

本文件沒有特定需求。

採用元件

本文中的資訊係根據以下軟體和硬體版本：

- 兩台傳真機
- 執行Cisco IOS®軟體版本12.4(4)T1的VG224
- 運行Cisco IOS軟體版本12.3(11)T8的Cisco 3745路由器

本文中的資訊是根據特定實驗室環境內的裝置所建立。文中使用到的所有裝置皆從已清除（預設）的組態來啟動。如果您的網路正在作用，請確保您已瞭解任何指令可能造成的影響。

慣例

如需文件慣例的詳細資訊，請參閱[思科技術提示慣例。](#)

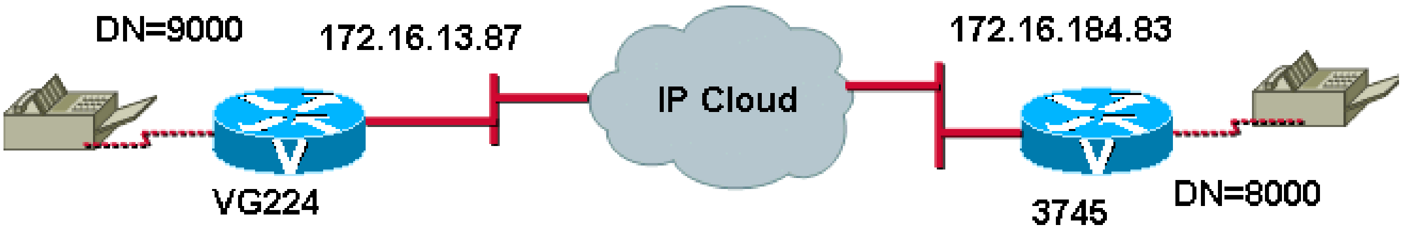
設定

本節提供用於設定本文件中所述功能的資訊。

註：使用[Command Lookup Tool](僅限[註冊]客戶)查詢有關本文檔中使用的命令的更多資訊。

網路圖表

本檔案會使用以下網路設定：



組態

本檔案會使用以下設定：

- VG224
- 思科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
!
```

思科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
```

驗證

目前沒有適用於此組態的驗證程序。

疑難排解

使用本節內容，對組態進行疑難排解。

輸出直譯器工具(僅供已註冊客戶使用)(OIT)支援某些**show**命令。使用OIT檢視**show**命令輸出的分析。

附註：使用 **debug** 指令之前，請先參閱有關 Debug 指令的重要資訊。

以下是**debug ccsip messages**命令的輸出：

```
!--- 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

相關資訊

- SDP RFC 2327
- SIP RFC 3261
- 語音技術支援
- 語音和整合通訊產品支援
- Cisco IP電話故障排除
- 技術支援與文件 - Cisco Systems