Risoluzione dei problemi relativi a una chiamata SIP tra due endpoint

Sommario

Introduzione

In questo documento viene fornita una configurazione di esempio di due fax per dimostrare come viene eseguita una chiamata SIP (Session Initiation Protocol) tra due gateway. In questo documento viene spiegato anche l'output del comando **debug ccsip messages** per la risoluzione dei problemi relativi agli errori delle chiamate SIP.

Prerequisiti

Requisiti

Nessun requisito specifico previsto per questo documento.

Componenti usati

Le informazioni fornite in questo documento si basano sulle seguenti versioni software e hardware:

- Due fax
- VG224 con software Cisco IOS® versione 12.4(4)T1
- Router Cisco 3745 con software Cisco IOS versione 12.3(11)T8

Le informazioni discusse in questo documento fanno riferimento a dispositivi usati in uno specifico ambiente di emulazione. Su tutti i dispositivi menzionati nel documento la configurazione è stata ripristinata ai valori predefiniti. Se la rete è operativa, valutare attentamente eventuali conseguenze derivanti dall'uso dei comandi.

Convenzioni

Per ulteriori informazioni sulle convenzioni usate, consultare il documento Cisco sulle convenzioni nei suggerimenti tecnici.

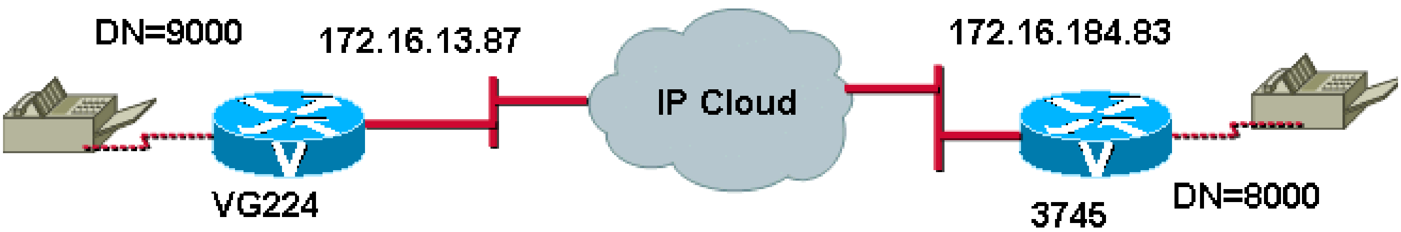
Configurazione

In questa sezione vengono presentate le informazioni necessarie per configurare le funzionalità descritte più avanti nel documento.

Nota: per ulteriori informazioni sui comandi menzionati in questo documento, usare lo strumento di ricerca dei comandi (solo utenti registrati).

Esempio di rete

Nel documento viene usata questa impostazione di rete:



Configurazioni

Nel documento vengono usate queste configurazioni:

- 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
```

Verifica

Attualmente non è disponibile una procedura di verifica per questa configurazione.

Risoluzione dei problemi

Utilizzare questa sezione per risolvere i problemi relativi alla configurazione.

Lo strumento Output Interpreter (solo utenti registrati) (OIT) supporta alcuni comandi **show**. Usare l'OIT per visualizzare un'analisi dell'output del comando **show**.

Nota: consultare le informazioni importanti sui comandi di debug prima di usare i comandi di **debug**.

Di seguito viene riportato l'output del comando **debug ccsip messages**:

```
!--- 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
```

Informazioni correlate

- SDP RFC 2327
- SIP RFC 3261
- Supporto alla tecnologia vocale
- Supporto ai prodotti voce e Unified Communications
- Risoluzione dei problemi di Cisco IP Telephony
- Documentazione e supporto tecnico – Cisco Systems