# Solución de problemas de una llamada SIP entre dos terminales

## Contenido



## Introducción

Este documento proporciona una configuración de ejemplo de dos máquinas de fax para demostrar cómo ocurre una llamada SIP (Session Initiation Protocol) entre dos gateways. Este documento también proporciona una explicación de la salida del comando debug ccsip messages para el Troubleshooting de las fallas de llamada SIP.

## Prerequisites

### Requirements

No hay requisitos específicos para este documento.

### Componentes Utilizados

La información que contiene este documento se basa en las siguientes versiones de software y hardware.

- Dos máquinas de fax
- VG224 que ejecuta Cisco IOS® Software Release 12.4(4)T1
- Cisco 3745 Router que ejecuta 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.

## Convenciones

Consulte Convenciones de Consejos TécnicosCisco para obtener más información sobre las convenciones del documento.

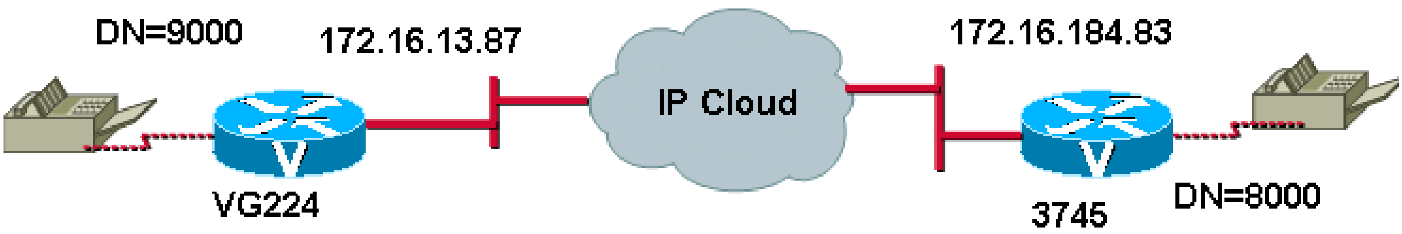
# Configurar

En esta sección encontrará la información para configurar las funciones descritas en este documento.

**Nota:** Use la Command Lookup Tool (sólo clientes registrados) para obtener más información sobre los comandos utilizados en este documento.

## Diagrama de la red

En este documento, se utiliza esta configuración de red:




## Configuraciones

En este documento, se utilizan estas configuraciones:

- VG224
- 3745 de Cisco

| VG224 |
|---|
| <pre>vg224#**show run**<br>Building configuration...<br>!<br>voice call send-alert<br>voice rtp send-recv<br>!<br>voice service pots<br>!<br>voice service voip<br> fax protocol t38 ls-redundancy 0 hs-redundancy 0<br>fallback cisco</pre> |

```
 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
```

## Verificación

Actualmente, no hay un procedimiento de verificación disponible para esta configuración.

## Troubleshoot

Use esta sección para resolver problemas de configuración.

La herramienta Output Interpreter Tool (clientes registrados solamente) (OIT) soporta ciertos comandos show. Utilice la OIT para ver un análisis del resultado del comando show.

**Nota:** Consulte Información Importante sobre Comandos Debug antes de utilizar los comandos **debug**.

Este es el resultado 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
```

## Información Relacionada

- RFC 2327 de SDP
- SIP RFC 3261
- Soporte de tecnología de voz
- Soporte de Productos de Voice and Unified Communications
- Troubleshooting de Cisco IP Telephony
- Soporte Técnico y Documentación - Cisco Systems