



Session Initiation Protocol Gateway Call Flows and Compliance Information

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About this Guide

This preface describes the objectives, audience, organization, and conventions of the *Session Initiation Protocol Gateway Call Flows and Compliance Information*.

This preface contains the following information:

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- Who Should Use This Guide, pageiii
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Overview

The *Session Initiation Protocol Gateway Call Flows and Compliance Information* provide call flows reflective of the Session Initiation Protocol (SIP) implementation on Cisco gateways. This document also includes RFC 2543 compliance information.

Who Should Use This Guide

Network engineers, system administrators, or telecommunication engineers should use this guide as a reference when implementing SIP in a network.

Document Objectives

The *Session Initiation Protocol Gateway Call Flows and Compliance Information* provides reference information for implementing SIP in a Voice-over-IP (VoIP) network.

It is not the intent of this document to provide information on how to implement a SIP VoIP network. For information on implementing a SIP VoIP network, refer to the documents listed in the “Related Documentation” section on page v.

Document Organization

This administrator guide is divided into the following chapters and appendixes:

- Chapter 1, “SIP Messages and Methods Overview” provides an overview of SIP messages and methods.
- Chapter 2, “Successful Call Flow Scenarios” provides illustrations and descriptions of call flows for successful calls.
- Chapter 3, “Call Flow Scenarios for Failed Calls” provides illustrations and descriptions of call flows for failed calls.
- Appendix A, “SIP Gateway Support for Third Party Call Control” describes the SIP Gateway Support for Third-Party Call Control feature that is introduced in Cisco IOS Release 12.2(1)T.
- Appendix B, “SIP Diversion Header Implementation for Redirecting Number” describes the SIP Diversion Header Implementation for Redirecting Number feature that is introduced in Cisco IOS Release 12.2(1)T.
- Appendix C, “Troubleshooting Tips for Call Flow Scenarios” describes the **debugccsipmessages** command traces the SIP messages exchanges between the SIP User Agent client and the gateway in the Cisco IOS Release 12.2(2)XB.

Document Conventions

Table 1 describes the conventions that might be used in this document.

Table 1 Document Conventions Guide

Convention	Description
boldface	Commands and keywords.
<i>italic</i>	Command input that is supplied by you.
[]	Keywords or arguments that appear within square brackets are optional.
{ x x x }	A choice of keywords (represented by x) appears in braces separated by vertical bars. You must select one.
^ or Ctrl	Represent the key labeled <i>Control</i> . For example, when you read ^D or <i>Ctrl-D</i> , you should hold down the Control key while you press the D key.
screen font	Examples of information displayed on the screen.
boldface screen font	Examples of information that you must enter.

Table 1 Document Conventions Guide (continued)

Convention	Description
< >	Nonprinting characters, such as passwords, appear in angled brackets.
[]	Default responses to system prompts appear in square brackets.

Related Documentation

The following is a list of related Cisco SIP VoIP publications. For more information about implementing a SIP VoIP network refer to the following publications:

- *Session Initiation for VoIP on Cisco Access Platforms*

The following is a list of Cisco VoIP publications that provide information about implementing a VoIP network:

- *Service Provider Features for Voice over IP*
- *Cisco IOS IP and IP Routing Configuration Guide*
- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2

Obtaining Documentation

These sections explain how to obtain documentation from Cisco Systems.

World Wide Web

You can access the most current Cisco documentation on the World Wide Web at this URL:

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Documentation CD-ROM

Cisco documentation and additional literature are available in a Cisco Documentation CD-ROM package, which is shipped with your product. The Documentation CD-ROM is updated monthly and may be more current than printed documentation. The CD-ROM package is available as a single unit or through an annual subscription.

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The Cisco Technical Assistance Center (TAC) is available to all customers who need technical assistance with a Cisco product, technology, or solution. Two levels of support are available: the Cisco TAC WebSite and the Cisco TAC Escalation Center.

Cisco TAC inquiries are categorized according to the urgency of the issue:

- Priority level 4 (P4)—You need information or assistance concerning Cisco product capabilities, product installation, or basic product configuration.
- Priority level 3 (P3)—Your network performance is degraded. Network functionality is noticeably impaired, but most business operations continue.
- Priority level 2 (P2)—Your production network is severely degraded, affecting significant aspects of business operations. No workaround is available.
- Priority level 1 (P1)—Your production network is down, and a critical impact to business operations will occur if service is not restored quickly. No workaround is available.

The Cisco TAC resource that you choose is based on the priority of the problem and the conditions of service contracts, when applicable.

Cisco TAC Web Site

You can use the Cisco TAC Web Site to resolve P3 and P4 issues yourself, saving both cost and time. The site provides around-the-clock access to online tools, knowledge bases, and software. To access the Cisco TAC Web Site, go to this URL:

<http://www.cisco.com/tac>

All customers, partners, and resellers who have a valid Cisco service contract have complete access to the technical support resources on the Cisco TAC Web Site. The Cisco TAC Web Site requires a Cisco.com login ID and password. If you have a valid service contract but do not have a login ID or password, go to this URL to register:

<http://www.cisco.com/register/>

If you are a Cisco.com registered user, and you cannot resolve your technical issues by using the Cisco TAC Web Site, you can open a case online by using the TAC Case Open tool at this URL:

<http://www.cisco.com/tac/caseopen>

If you have Internet access, we recommend that you open P3 and P4 cases through the Cisco TAC WebSite.

Cisco TAC Escalation Center

The Cisco TAC Escalation Center addresses priority level 1 or priority level 2 issues. These classifications are assigned when severe network degradation significantly impacts business operations. When you contact the TAC Escalation Center with a P1 or P2 problem, a Cisco TAC engineer automatically opens a case.

To obtain a directory of toll-free Cisco TAC telephone numbers for your country, go to this URL:

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Before calling, please check with your network operations center to determine the level of Cisco support services to which your company is entitled: for example, SMARTnet, SMARTnet Onsite, or Network Supported Accounts (NSA). When you call the center, please have available your service agreement number and your product serial number.



SIP Messages and Methods Overview

This chapter provides a brief overview of SIP messages and methods. This chapter contains information about the following:

- Format, page 1-1
- Requests, page 1-1
- The Invitation Process, page 1-2

Format

All SIP messages are either requests from a server or client or responses to a request. The messages are formatted according to RFC 822, “Standard for the format of ARPA internet text messages.” For all messages, the general format is:

- A start line
- One or more header fields
- An empty line
- A message body (optional)

Each line must end with a carriage return-line feed (CRLF).

Requests

SIP uses six types (methods) of requests:

- INVITE—Indicates a user or service is being invited to participate in a call session.
- ACK—Confirms that the client has received a final response to an INVITE request.
- BYE—Terminates a call and can be sent by either the caller or the callee.
- CANCEL—Cancels any pending searches but does not terminate a call that has already been accepted.
- OPTIONS—Queries the capabilities of servers.
- REGISTER—Registers the address listed in the To header field with a SIP server.

Responses

The following types of responses are used by SIP and generated by the Cisco SIP Proxy Server:

- SIP 1xx—Informational Responses
- SIP 2xx—Successful Responses
- SIP 3xx—Redirection Responses
- SIP 4xx—Client Failure Responses
- SIP 5xx—Server Failure Responses
- SIP 6xx—Global Failure Responses

The Registration Process

A registration occurs when a client needs to inform a proxy or redirect server of its location. During this process, the client sends a REGISTER request to the proxy or redirect server and includes the address (or addresses) at which it can be reached.

The Invitation Process

An invitation occurs when one SIP end point (user A) “invites” another SIP endpoint (user B) to join in a call. During this process, user A sends an INVITE message requesting that user B join a particular conference or establish a two-party conversation. If user B wants to join the call, it sends an affirmative response (SIP 2xx). Otherwise, it sends a failure response (SIP 4xx). Upon receiving the response, user A acknowledges the response with an ACK message. If user A no longer wants to establish this conference, it sends a BYE message instead of an ACK message.

SIP is a new protocol developed by the Internet Engineering Task Force (IETF) Multiparty Multimedia Session Control (MMUSIC) Working Group as an alternative to the ITU-T H.323 specification. SIP is defined by RFC 2543 and is used for multimedia call session setup and control over IP networks.



Successful Call Flow Scenarios

This chapter describes call flows for the following scenarios, which illustrate successful calls:

- SIP Gateway-to-SIP Gateway Call, page 2-1
- SIP Gateway-to-SIP Gateway Call via SIP Redirect Server, page 2-5
- SIP Gateway-to-SIP Gateway Call via SIP Proxy Server, page 2-9
- SIP Gateway-to-SIP IP Phone Calls
 - SIP Gateway-to-SIP IP Phone Calls, page 2-17
 - SIP Gateway-to-SIP IP Phone—Call Setup and Call Hold, page 2-19
 - SIP Gateway-to-SIP IP Phone—Call Setup and Transfer, page 2-22

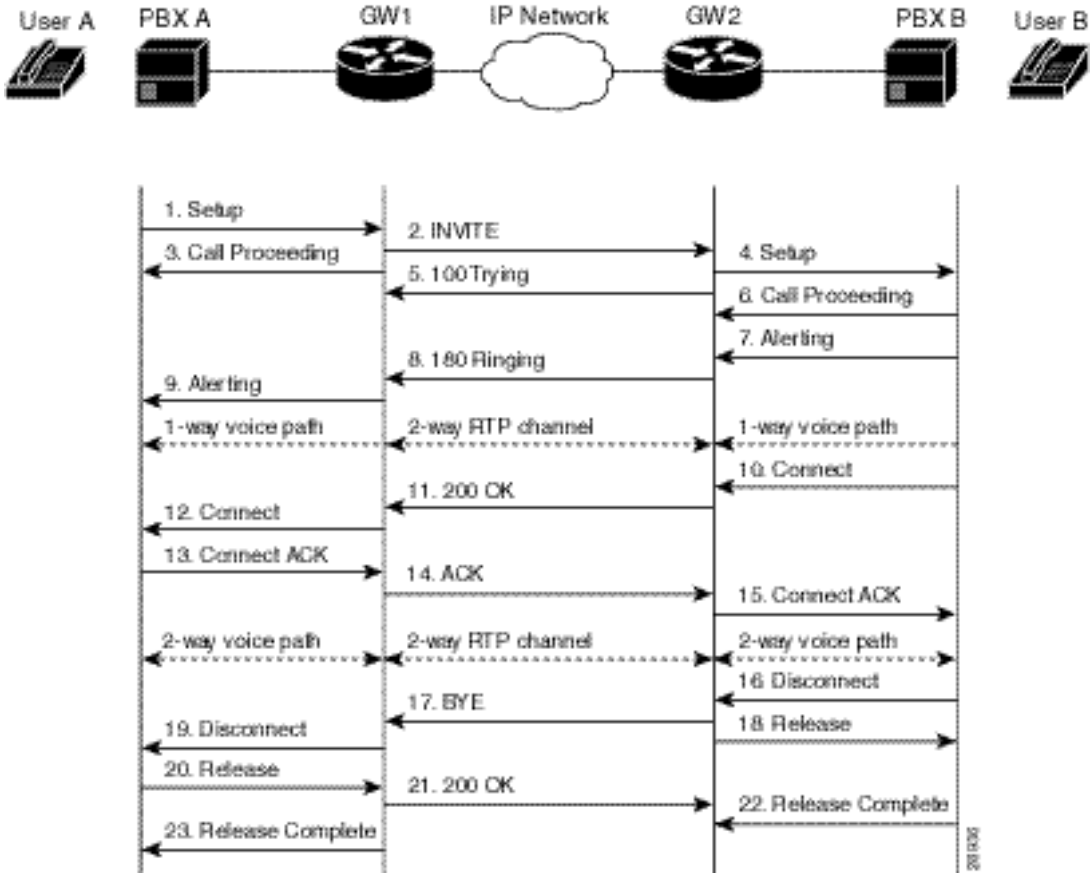
SIP Gateway-to-SIP Gateway Call

Figure 1 illustrates a successful gateway-to-gateway call setup and disconnect. In this call flow scenario, the two end users are User A and User B. User A is located at PBX A. PBX A is connected to SIP gateway 1 via a T1/E1. User B is located at PBX B. PBX B is connected to SIP gateway 2 via a T1/E1. User B's phone number is 555-0002. SIP gateway 1 is connected to SIP gateway 2 over an IP network.

The call flow scenario is as follows:

1. User A calls User B.
2. User B answers the call.
3. User B hangs up.

Figure 1 SIP Gateway-to-SIP Gateway Call



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—SIP Gateway 1 to SIP Gateway 2	<p>SIP gateway 1 sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter indicates that the Request-URI address is a telephone number rather than a user name. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates a Call Setup with User B via PBXB.
5	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
6	Call Proceeding—PBX B to SIP Gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
7	Alerting—PBX B to SIP Gateway 2	PBX B locates User B and sends an Alert message to SIP gateway 2. User B’s phone begins ringing.
8	180 Ringing—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that SIP gateway 2 has located, and is trying to alert, User B.
9	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to User A via PBX A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response from SIP gateway 2. User A hears the ringback tone that indicates that User B is being alerted.
<p>At this point, a one-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>		
10	Connect—PBX B to SIP Gateway 2	User B answers phone. PBX B sends a Connect message to SIP gateway 2. The Connect message notifies SIP gateway 2 that the connection has been made.

Step	Action	Description
11	200 OK—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made. If User B supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400Bad Request response with a 304 Warning header field.
12	Connect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
13	Connect ACK—PBX A to SIP Gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
14	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a an ACK to SIP gateway 2. The ACK confirms that SIP gateway 1 has received the 200 OK response from SIP gateway 2.
15	Connect ACK—SIP Gateway 2 to PBX B	SIP gateway 2 acknowledges PBX B's Connect message. The call session is now active over a two-way voice path via Real Time Transport Protocol (RTP).
At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.		
16	Disconnect—PBX B to SIP Gateway 2	Once User B hangs up, PBX B sends a Disconnect message to SIP gateway 2. The Disconnect message starts the call session termination process.
17	BYE—SIP Gateway 2 to SIP Gateway 1	SIP Gateway 2 sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
18	Release—SIP Gateway 2 to PBX B	SIP gateway 2 sends a Release message to PBX B.
19	Disconnect—SIP Gateway 1 to PB X A	SIP gateway 1 sends a Disconnect message to PBX A.
20	Release—PBX A to SIP Gateway 1	PBX A sends a Disconnect message to SIP gateway 1.
21	200 OK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.
22	Release Complete—PBX B to SIP Gateway 2	PBX B sends a Release Complete message to SIP gateway 2.
23	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the session is terminated.

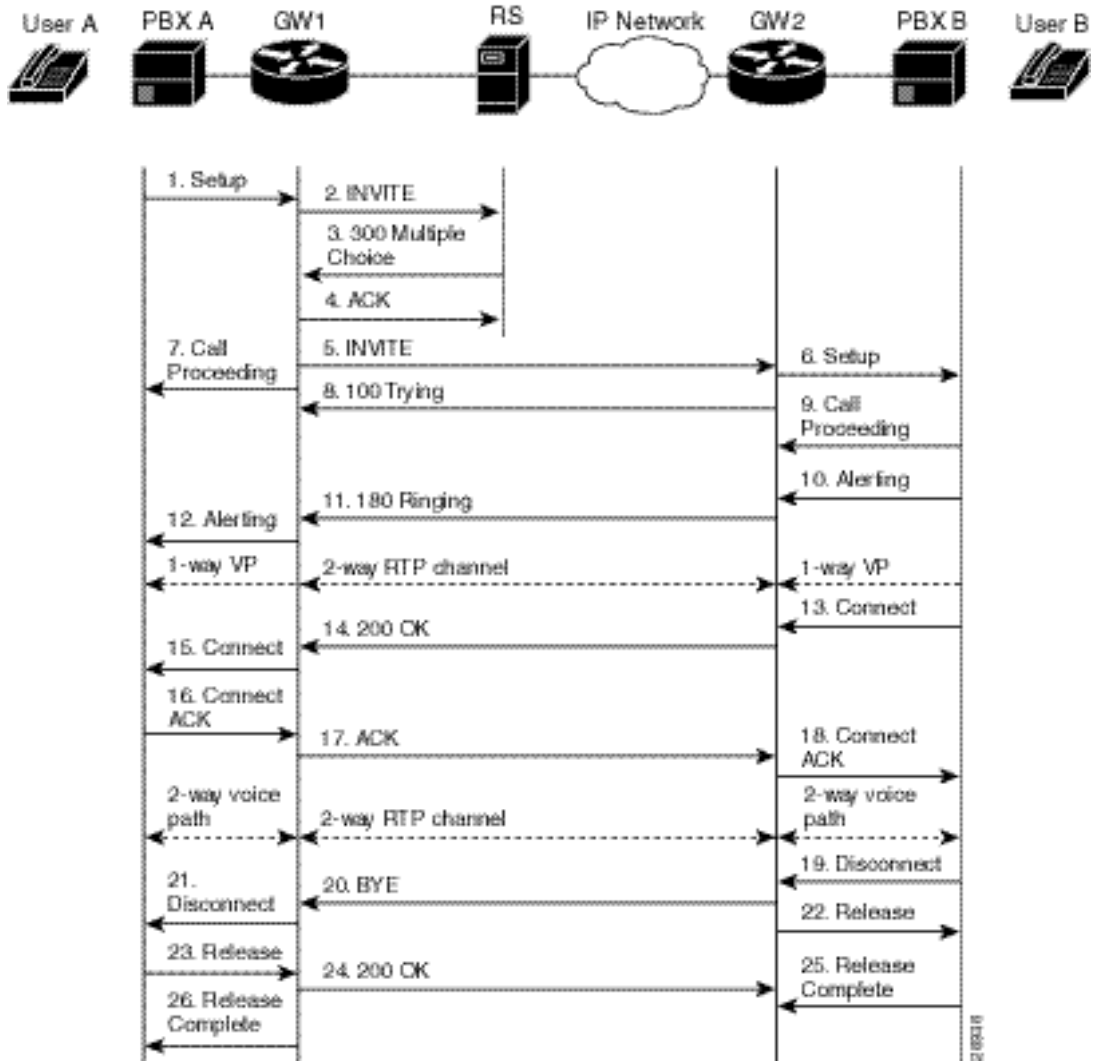
SIP Gateway-to-SIP Gateway Call via SIP Redirect Server

Figure 2 illustrates a successful gateway-to-gateway call setup and disconnect via a SIP redirect server. In this scenario, the two end users are identified as User A and User B. User A is located at PBX A. PBX A is connected to SIP gateway 1 via a T1/E1. SIP gateway 1 is using a SIP redirect server. User B is located at PBX B. PBX B is connected to SIP gateway 2 via a T1/E1. User B's phone number is 555-0002. SIP gateway 1 is connected to SIP gateway 2 over an IP network.

The call flow scenario is as follows:

1. User A calls User B via SIP gateway 1 using a SIP redirect server.
2. User B answers the call.
3. User B hangs up.

Figure2 SIP Gateway-to-SIP Gateway Call via SIP Redirect Server



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—SIP Gateway 1 to SIP Redirect Server	<p>SIP gateway 1 sends an INVITE request to the SIP redirect server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a user name. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	300 Multiple Choice—SIP Redirect Server to SIP Gateway 1	The SIP redirect server sends a 300 Multiple Choice response to SIP gateway 1. The 300 Multiple Choice response indicates that the SIP redirect server accepted the INVITE request, contacted a location server with all or part of User B’s SIP URL, and the location server provided a list of alternative locations where User B might be located. The SIP redirect server returns these possible addresses to SIP gateway 1 in the 300 Multiple Choice response.
4	ACK—SIP Gateway 1 to SIP Redirect Server	SIP gateway 1 acknowledges the 300 Multiple Choice response with an ACK.
5	INVITE—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a new INVITE request to SIP gateway 2. The new INVITE request includes the first contact listed in the 300 Multiple Choice response as the new address for User B, a higher transaction number in the CSeq field, and the same Call-ID as the first INVITE request.
6	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates a Call Setup with User B via PBXB.
7	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
8	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located.
9	Call Proceeding—PBX B to SIP Gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
10	Alerting—PBX B to SIP Gateway 2	PBX B locates User B and sends an Alert message to SIP gateway 2. User B’s phone begins to ring.

Step	Action	Description
11	180 Ringing—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that SIP gateway 2 has located, and is trying to alert, User B.
12	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to PBX A. User A hears ringback tone.
At this point, a one-way voice path is established between SIP gateway 1 and PBXA and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.		
13	Connect—PBX B to SIP Gateway 2	User B answers phone. PBX B sends a Connect message to SIP gateway 2. The Connect message notifies SIP gateway 2 that the connection has been made.
14	200 OK—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made. If User B supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.
15	Connect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
16	Connect ACK—PBX A to SIP Gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
17	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that the 200 OK response has been received. The call is now in progress over a two-way voice path via RTP.
18	Connect ACK—SIP Gateway 2 to PBX B	SIP gateway 2 acknowledges PBX B's Connect message.
At this point, a two-way voice path is established between SIP gateway 1 and PBXA and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.		
19	Disconnect—PBX B to SIP Gateway 2	Once User B hangs up, PBX B sends a Disconnect message to SIP gateway 2. The Disconnect message starts the call session termination process.
20	BYE—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
21	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
22	Release—SIP Gateway 2 to PBX B	SIP gateway 2 sends a Release message to PBX B.
23	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
24	200 OK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.

Step	Action	Description
25	Release Complete—PBX B to SIP Gateway 2	PBX B sends a Release Complete message to SIP gateway 2.
26	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the session is terminated.

SIP Gateway-to-SIP Gateway Call via SIP Proxy Server

Figure3 and Figure4 illustrate a successful gateway-to-gateway call setup and disconnect via a proxy server. In these scenarios, the two end users are User A and User B. User A is located at PBX A. PBX A is connected to SIP Gateway 1 via a T1/E1. SIP Gateway 1 is using a proxy server. SIP Gateway 1 is connected to SIP Gateway 2 over an IP network. User B is located at PBX B. PBX B is connected to SIP Gateway 2 (a SIP gateway) via a T1/E1. User B's phone number is 555-0002.

In the scenario illustrated by Figure3, the record route feature is enabled on the proxy server. In the scenario illustrated by Figure4, record route is disabled on the proxy server.

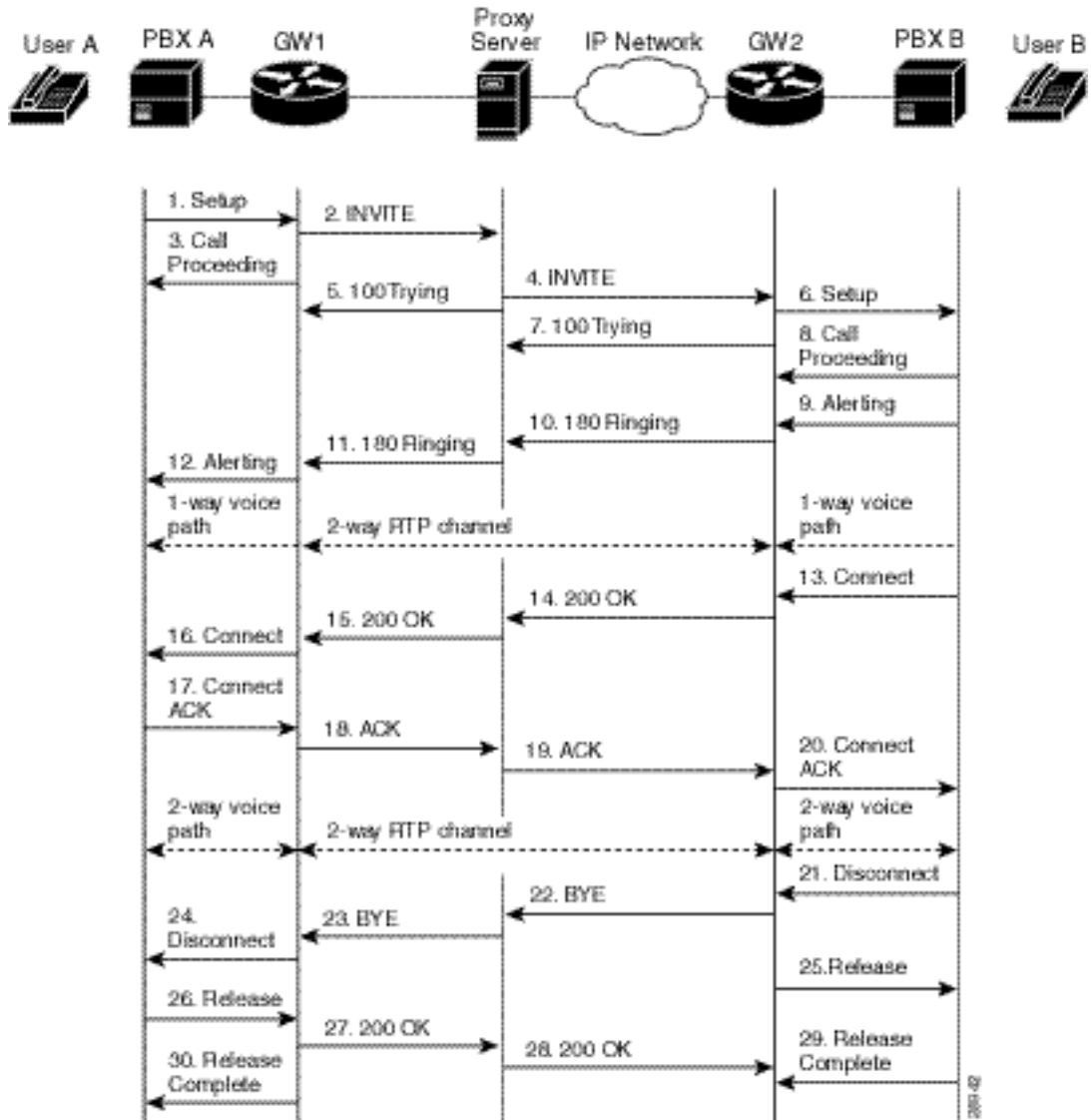
When record route is enabled, the proxy server adds the Record-Route header to the SIP messages to ensure that it is in the path of subsequent SIP requests for the same call leg. The Record-Route field contains a globally reachable Request-URI that identifies the proxy server. When record route is enabled, each proxy server adds its Request-URI to the beginning of the list.

When record route is disabled, SIP messages flow directly through the SIP gateways once a call has been established.

The call flow is as follows:

1. User A calls User B via SIP gateway 1 using a proxy server.
2. User B answers the call.
3. User B hangs up.

Figure3 SIP Gateway-to-SIP Gateway Call via SIP Proxy Server with Record Route Enabled



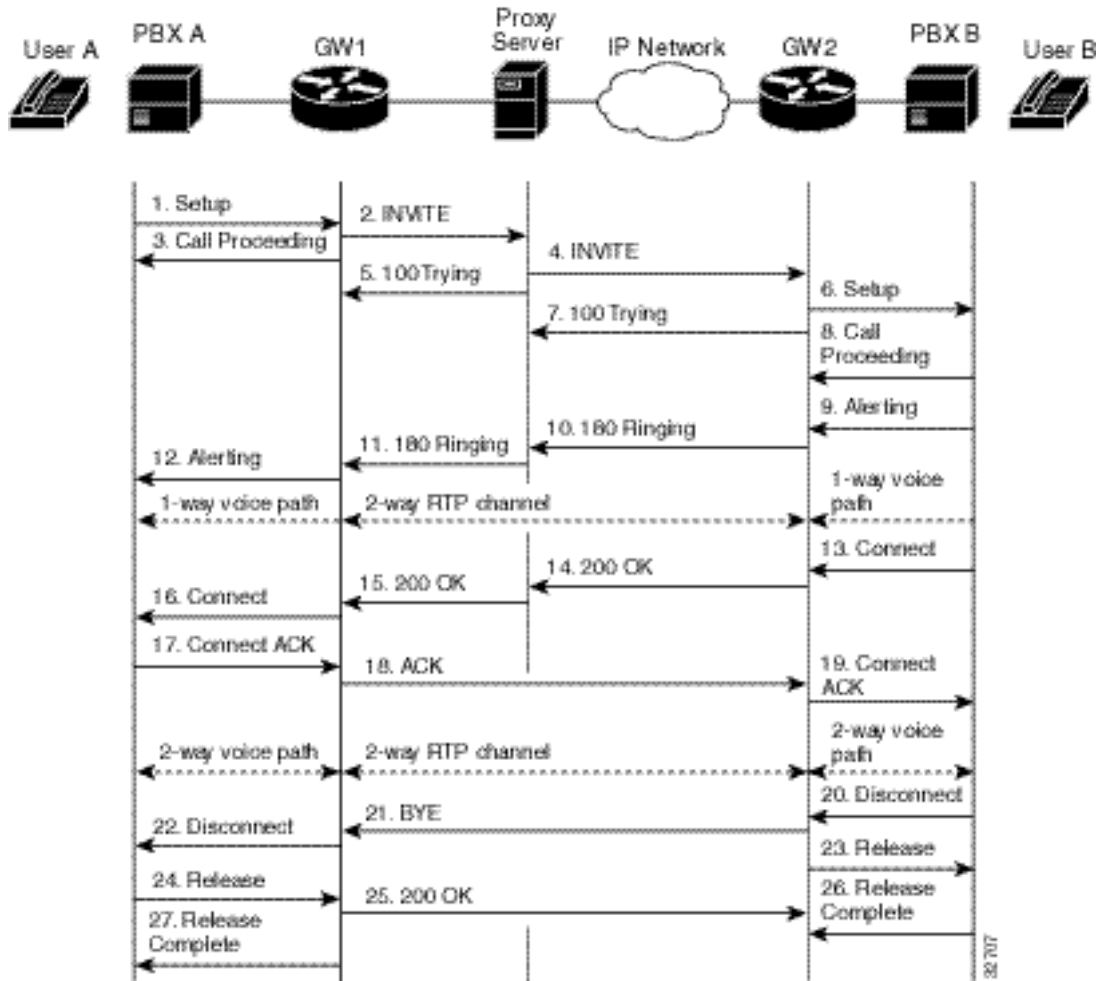
Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—SIP Gateway 1 to Proxy Server	<p>SIP gateway 1 sends an INVITE request to the SIP proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a user name. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	INVITE—SIP Proxy Server to SIP Gateway 2	<p>The SIP proxy server checks whether it’s own address is contained in the Via field (to prevent loops), directly copies the To, From, Call-ID, and Contact fields from the request it received from SIP gateway 1, changes the Request-URI to indicate the server to which it intends to send the INVITE request, and then sends a new INVITE request to SIP gateway 2.</p> <p>In the INVITE request, the SIP proxy server also adds the Record-Route header to ensure that it is in the path of subsequent SIP requests for the same call leg.</p> <p>In the Record-Route field, the SIP proxy server adds its Request-URI.</p>
5	100 Trying—SIP Proxy Server to SIP Gateway 1	The SIP proxy server sends a 100 Trying response to SIP gateway 1.
6	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from the SIP proxy server and initiates a Call Setup with UserB via PBXB.
7	100 Trying—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 sends a 100 Trying response to the SIP proxy server. The SIP proxy server might or might not forward the 100 Trying response to SIP gateway 1.
8	Call Proceeding—PBX B to SIP Gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
9	Alerting—PBX B to SIP Gateway 2	PBX B locates User B and sends an Alert message to SIP gateway 2. User B’s phone begin to ring.
10	180 Ringing—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 sends a 180 Ringing response to the SIP proxy server.

Step	Action	Description
11	180 Ringing—SIP Proxy Server to SIP Gateway 1	The SIP proxy server forwards the 180 Ringing response to SIP gateway 1.
12	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to User A via PBX A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response. User A hears the ringback tone that indicates that User B is being alerted.
At this point, a one-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.		
13	Connect—PBX B to SIP Gateway 2	User B answers the phone. PBX B sends a Connect message to SIP gateway 2. The connect message notifies SIP gateway 2 that the connection has been made.
14	200 OK—SIP Gateway 2 to SIP Proxy Server	<p>SIP gateway 2 sends a 200 OK response to the SIP proxy server. The 200 OK response notifies the SIP proxy server that the connection has been made.</p> <p>In the 200 OK response, the SIP gateway 2 adds the Record-Route header (received in the INVITE request) to ensure that it is in the path of subsequent SIP requests for the same call leg.</p> <p>If User B supports the media capability advertised in the INVITE message sent by the SIP proxy server, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.</p> <p>The SIP proxy server must forward 200 OK responses.</p>
15	200 OK—SIP Proxy Server to SIP Gateway 1	The SIP proxy server forwards the 200 OK response that it received from SIP gateway 2 to SIP gateway 1.
16	Connect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
17	Connect ACK—PBX A to SIP Gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
18	ACK—SIP Gateway 1 to SIP Proxy Server	SIP gateway 1 sends an ACK to the SIP proxy server. The ACK confirms that SIP gateway 1 has received the 200 OK response from the SIP proxy server.
19	ACK—SIP Proxy Server to SIP Gateway 2	<p>Depending on the values in the To, From, CSeq, and Call-ID field, the SIP proxy server might process the ACK locally or proxy it. If the fields in the ACK match those in previous requests processed by the SIP proxy server, the server proxies the ACK. If there is no match, the ACK is proxied as if it were an INVITE request.</p> <p>The SIP proxy server forwards SIP gateway 1's ACK response to SIP gateway 2.</p>
20	Connect ACK—SIP Gateway 2 to PBX B	<p>SIP gateway 2 acknowledges PBX B's Connect message. The call session is now active.</p> <p>The 2-way voice path is established directly between SIP gateway 1 and SIP gateway 2; not via the SIP proxy server.</p>
At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.		
21	Disconnect—PBX B to SIP Gateway 2	After the call is completed, PBX B sends a Disconnect message to SIP gateway2. The Disconnect message starts the call session termination process.

Step	Action	Description
22	BYE—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 sends a BYE request to the SIP proxy server. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains UserB's SIP URL.
23	BYE—SIP Proxy Server to SIP Gateway 1	The SIP proxy server forwards the BYE request to SIP gateway 1.
24	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
25	Release—SIP Gateway 2 to PBX B	After the call is completed, SIP gateway 2 sends a Release message to PBX B.
26	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
27	200 OK—SIP Gateway 1 to SIP Proxy Server	SIP gateway 1 sends a 200 OK response to the SIP proxy server. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.
28	200 OK—SIP Proxy Server to SIP Gateway 2	The SIP proxy server forwards the 200 OK response to SIP gateway 2.
29	Release Complete—PBX B to SIP Gateway 2	PBX B sends a Release Complete message to SIP gateway 2.
30	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session is terminated.

Figure4 SIP Gateway-to-SIP Gateway Call via a Proxy Server with Record Route Disabled



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—SIP Gateway 1 to SIP Proxy Server	<p>SIP gateway 1 sends an INVITE request to the SIP proxy server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a user name. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	INVITE—SIP Proxy Server to SIP Gateway 2	The SIP proxy server checks whether it’s own address is contained in the Via field (to prevent loops), directly copies the To, From, Call-ID, and Contact fields from the request it received from SIP gateway 1, changes the Request-URI to indicate the server to which it intends to send the INVITE request, and then sends a new INVITE request to SIP gateway 2.
5	100 Trying—SIP Proxy Server to SIP Gateway 1	The SIP proxy server sends a 100 Trying response to SIP gateway 1.
6	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from the SIP proxy server and initiates a Call Setup with UserB via PBXB.
7	100 Trying—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 sends a 100 Trying response to the SIP proxy server. The SIP proxy server might or might not forward the 100 Trying response to SIP gateway 1.
8	Call Proceeding—PBX B to SIP Gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
9	Alerting—PBX B to SIP Gateway 2	PBX B locates User B and sends an Alert message to SIP gateway 2. User B’s phone begin to ring.
10	180 Ringing—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 sends a 180 Ringing response to the SIP proxy server.
11	180 Ringing—SIP Proxy Server to SIP Gateway 1	The SIP proxy server forwards the 180 Ringing response to SIP gateway 1.

Step	Action	Description
12	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to User A via PBX A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response. User A hears the ringback tone that indicates that UserB is being alerted.
<p>At this point, a one-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>		
13	Connect—PBX B to SIP Gateway 2	User B answers the phone. PBX B sends a Connect message to SIP gateway 2. The connect message notifies SIP gateway 2 that the connection has been made.
14	200 OK—SIP Gateway 2 to SIP Proxy Server	<p>SIP gateway 2 sends a 200 OK response to the SIP proxy server. The 200 OK response notifies the SIP proxy server that the connection has been made.</p> <p>If User B supports the media capability advertised in the INVITE message sent by the SIP proxy server, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.</p> <p>The SIP proxy server must forward 200 OK responses.</p>
15	200 OK—SIP Proxy Server to SIP Gateway 1	The SIP proxy server forwards the 200 OK response that it received from SIP gateway 2 to SIP gateway 1.
16	Connect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
17	Connect ACK—PBX A to SIP Gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
18	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that SIP gateway 1 has received the 200 OK response from the SIP proxy server.
19	Connect ACK—SIP Gateway 2 to PBX B	<p>SIP gateway 2 acknowledges PBX B's Connect message. The call session is now active.</p> <p>The 2-way voice path is established directly between SIP gateway 1 and SIP gateway 2; not via the SIP proxy server.</p>
<p>At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>		
20	Disconnect—PBX B to SIP Gateway 2	After the call is completed, PBX B sends a Disconnect message to SIP gateway2. The Disconnect message starts the call session termination process.
21	BYE—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
22	Disconnect—SIP Gateway 1 to PB X A	SIP gateway 1 sends a Disconnect message to PBX A.
23	Release—SIP Gateway 2 to PBX B	After the call is completed, SIP gateway 2 sends a Release message to PBX B.
24	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.

Step	Action	Description
25	200 OK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.
26	Release Complete—PBX B to SIP Gateway 2	PBX B sends a Release Complete message to SIP gateway 2.
27	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session is terminated.

SIP Gateway-to-SIP IP Phone Calls

This section describes the following SIP Gateway-to-SIP IP phone call flows:

- SIP Gateway-to-SIP IP Phone—Call Setup and Disconnect, page 2-17
- SIP Gateway-to-SIP IP Phone—Call Setup and Call Hold, page 2-19
- SIP Gateway-to-SIP IP Phone—Call Setup and Transfer, page 2-22

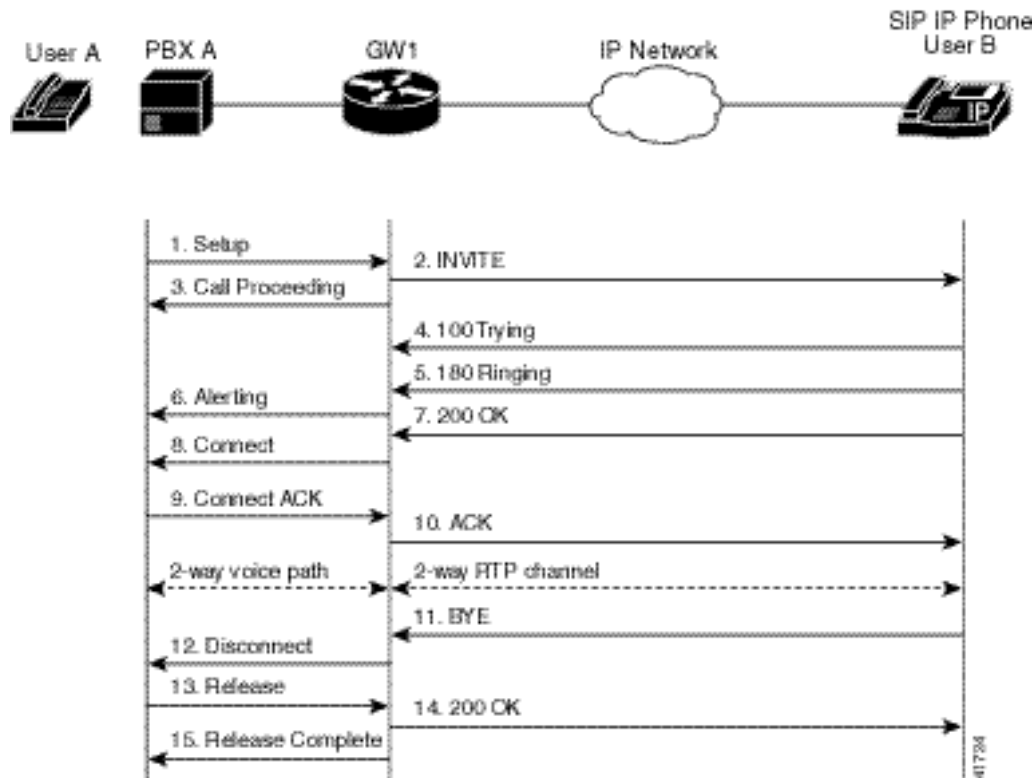
SIP Gateway-to-SIP IP Phone—Call Setup and Disconnect

Figure 5 illustrates a successful gateway-to-SIP IP phone call setup and disconnect. In this scenario, the two end users are User A and User B. User A is located at PBX A. PBX A is connected to SIP gateway 1 via a T1/E1. User B is located at a SIP IP phone. SIP gateway 1 is connected to the SIP IP phone over an IP network.

The call flow is as follows:

1. User A calls User B.
2. User B answers the call.
3. User B hangs up.

Figure5 SIP Gateway-to-SIP IP Phone—Call Setup and Disconnect



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP IP Phone	<p>SIP gateway 1 maps the SIP URL phone number to a dial-peer. The dial-peer includes the IP address and the port number of the SIP enabled entity to contact. SIP gateway 1 sends an INVITE request to the address it receives in the dial peer which, in this scenario, is the SIP IP phone.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The IP address of the SIP IP phone is inserted in the Request-URI field. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which the SIP gateway is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.

Step	Action	Description
4	100 Trying—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 100 Trying response to SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by the SIP IP phone.
5	180 Ringing—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that the user is being alerted.
6	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to User A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response from the SIP IP phone. User A hears the ringback tone that indicates that User B is being alerted.
7	200 OK—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made.
8	Connect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
9	Connect ACK—PBX A to SIP Gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
10	ACK—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 sends an ACK to the SIP IP phone. The ACK confirms that SIP gateway 1 has received the 200 OK response. The call session is now active.
At this point, a two-way voice path is established between SIP gateway 1 and PBX A. A two-way RTP channel is established between SIP gateway 1 and SIP IP phone B.		
11	BYE—SIP IP Phone to SIP Gateway 1	User B terminates the call session at his SIP IP phone and the phone sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call.
12	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
13	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
14	200 OK—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 sends a 200 OK response to the SIP IP phone. The 200 OK response notifies the phone that SIP gateway 1 has received the BYE request.
15	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session is terminated.

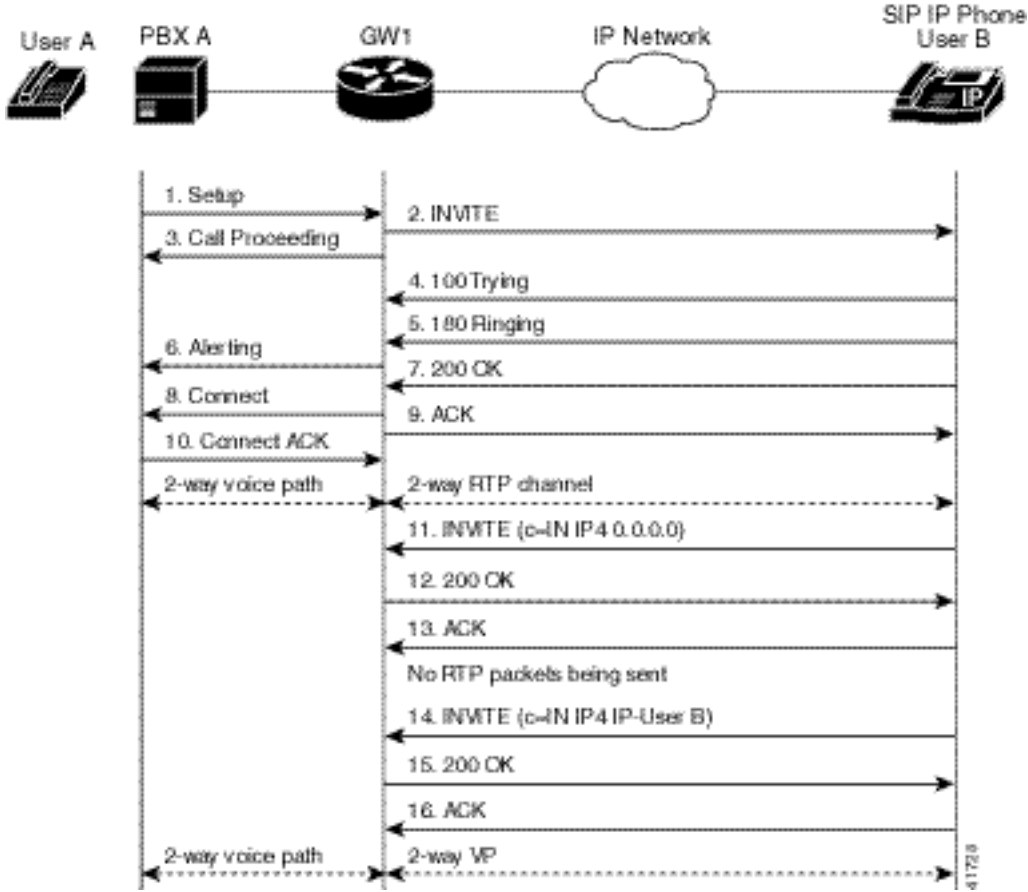
SIP Gateway-to-SIP IP Phone—Call Setup and Call Hold

Figure 6 illustrates a successful gateway-to-SIP IP phone call setup and call hold. In this scenario, the two end users are User A and User B. User A is located at PBX A. PBX A is connected to SIP gateway 1 via a T1/E1. User B is located at a SIP IP phone. SIP gateway 1 is connected to the SIP IP phone over an IP network.

The call flow is as follows:

- 1. User A calls User B.
- 2. User B answers the call.
- 3. User B puts User A on hold.
- 4. User B takes User A off hold.

Figure6 SIP Gateway-to-SIP IP Phone Call—Call Setup and Call Hold



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—SIP Gateway 1 to SIP IP Phone	<p>SIP gateway 1 maps the SIP URL phone number to a dial-peer. The dial-peer includes the IP address and the port number of the SIP enabled entity to contact. SIP gateway 1 sends an INVITE request to the address it receives in the dial peer which, in this scenario, is the SIP IP phone.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The IP address of the SIP IP phone is inserted in the Request-URI field. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which the SIP gateway is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	100 Trying—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 100 Trying response to SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by the SIP IP phone.
5	180 Ringing—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that the user is being alerted.
6	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to User A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response from the SIP IP phone. User A hears the ringback tone that indicates that User B is being alerted.
7	200 OK—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made.
8	Connect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
9	ACK—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 sends an ACK to the SIP IP phone. The ACK confirms that User A has received the 200 OK response. The call session is now active.
10	Connect ACK—PBX A to SIP Gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
<p>At this point, a two-way voice path is established between SIP gateway 1 and PBX A. A two-way RTP channel is established between SIP gateway 1 and SIP IP phone B.</p>		
11	INVITE—SIP IP Phone to SIP Gateway 1	User B puts User A on hold. The SIP IP phone sends an INVITE request to SIP gateway 1.
12	200 OK—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 sends a 200 OK response to the SIP IP phone. The 200 OK response notifies the SIP IP phone that the INVITE was successfully processed.
13	ACK—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends an ACK to SIP gateway 1. The ACK confirms that the SIP IP phone has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.

Step	Action	Description
The two-way RTP channel is torn down. The call is on hold.		
14	INVITE—SIP IP Phone to SIP Gateway 1	User B takes User A off hold. The SIP IP phone sends an INVITE request to SIP gateway 1.
15	200 OK—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 sends a 200 OK response to the SIP IP phone. The 200 OK response notifies the SIP IP phone that the INVITE was successfully processed.
16	ACK—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends an ACK to SIP gateway 1. The ACK confirms that the SIP IP phone has received the 200 OK response. The call session is now active.

At this point, a two-way voice path exists between SIP gateway 1 and PBX A and the two-way RTP channel is reestablished between SIP gateway 1 and SIP IP phone B.

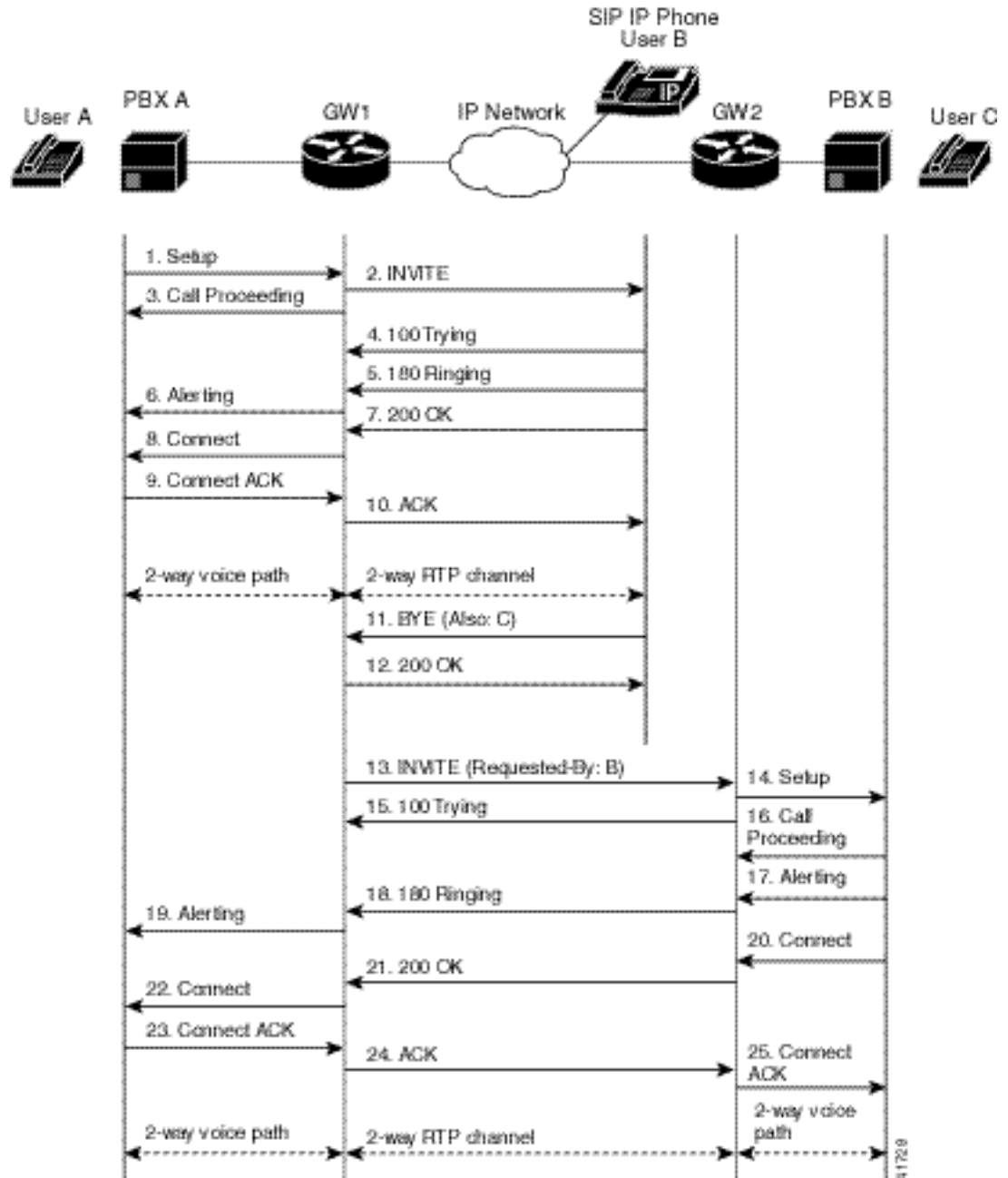
SIP Gateway-to-SIP IP Phone—Call Setup and Transfer

Figure7 illustrates a successful gateway-to-SIP IP phone call setup and call transfer without consultation. In this scenario, there are three end users: User A, User B, and User C. User A is located at PBX A. PBX A is connected to SIP gateway 1 via a T1/E1. User B is located at a SIP IP phone and is directly connected to the IP network. User C is located at PBX B. PBX B is connected to SIP gateway2 via a T1/E1. SIP gateway 1, SIP gateway 2, and the SIP IP phone are connected to one another over an IP network.

The call flow is as follows:

1. User A calls User B.
2. User B answers the call.
3. User B transfers User A's call to User C and then hangs up.
4. User C answers the call.

Figure7 SIP Gateway-to-SIP IP Phone Call—Call Setup and Transfer



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Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—SIP Gateway 1 to SIP IP Phone	<p>SIP gateway 1 maps the SIP URL phone number to a dial-peer. The dial-peer includes the IP address and the port number of the SIP enabled entity to contact. SIP gateway 1 sends an INVITE request to the address it receives in the dial peer which, in this scenario, is the SIP IP phone.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The IP address of the SIP IP phone is inserted in the Request-URI field. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which the SIP gateway is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	100 Trying—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 100 Trying response to SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by the SIP IP phone.
5	180 Ringing—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that the user is being alerted.
6	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to User A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response from the SIP IP phone. User A hears the ringback tone that indicates that User B is being alerted.
7	200 OK—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made.
8	Connect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
9	Connect ACK—PBX A to SIP Gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
10	ACK—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 sends an ACK to the SIP IP phone. The ACK confirms that SIP gateway 1 has received the 200 OK response. The call session is now active.

At this point, a two-way voice path is established between SIP gateway 1 and PBX A. A two-way RTP channel is established between SIP gateway 1 and SIP IP phone B.

Step	Action	Description
11	BYE—SIP IP Phone to SIP Gateway 1	<p>User B transfers User A's call to User C and then hangs up. The SIP IP phone sends a BYE request to SIP gateway 1. The BYE request includes the Also header. In this scenario, the Also header indicates that User C needs to be brought into the call while User B hangs up. This header distinguishes the call transfer BYE request from a normal disconnect BYE request.</p> <p>The Request-By header could be included in the BYE request, however, Cisco's implementation does not require the header to complete the transfer. If the Requested-By header is included, the INVITE sent to the transferred-to party will include the Requested-By header. If the Requested-By header is not included, the INVITE sent to the transferred-to party will not include the Requested-By header.</p>
12	200 OK—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 sends a 200 OK message to the SIP IP phone. The 200 OK response notifies the SIP IP phone that the BYE request has been received. The call session between User A and User B is now terminated.
13	INVITE—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an INVITE request to SIP gateway 2. In the INVITE request, a unique Call-ID is generated and the Requested-By field indicates that User B requested the call.
14	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates a Call Setup with User C via PBX B.
15	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that SIP gateway 2 has received the INVITE request but that User C has not yet been located.
16	Call Proceeding—PBX B to SIP Gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2. User C's phone begins to ring.
17	Alerting—PBX B to SIP Gateway 2	PBX B sends an Alert message to SIP gateway 2. The message indicates that the PBX is attempting to alert the user of the call (that is to say, the phone is ringing).
18	180 Ringing—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that SIP gateway 2 has located, and is trying to alert, User C.
19	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to PBX A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response from the SIP IP phone. User A hears the ringback tone that indicates that User C is being alerted.
20	Connect—PBX B to SIP Gateway 2	User C answers the phone. PBX B sends a Connect message to SIP gateway 2. The Connect message notifies SIP gateway 2 that the connection has been made.
21	200 OK—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made.</p> <p>If User C supports the media capability advertised in the INVITE message sent by User A, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User C does not support the media capability advertised by User A, it sends back a 400Bad Request response with a 304 Warning header field.</p>
22	Connect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.

Step	Action	Description
23	Connect ACK—PBX A to SIP Gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
24	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that SIP gateway 1 has received the 200 OK message from SIP gateway 2.
25	Connect ACK—SIP Gateway 2 to PBX B	SIP gateway 2 acknowledges PBX B's Connect message.

At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.



Call Flow Scenarios for Failed Calls

This chapter describes call flows for the following scenarios, which illustrate failed calls:

- SIP Gateway-to-SIP Gateway Calls
 - The Called User is Busy, page 3-2
 - The Called User Does Not Answer, page 3-3
 - A Client, Server, or Global Error Has Occurred, page 3-5
- SIP Gateway-to-SIP Gateway Calls via a SIP Redirect Server
 - The Called User is Busy, page 3-7
 - The Called User Does Not Answer, page 3-9
 - A Client, Server, or Global Error Has Occurred, page 3-12
- SIP Gateway-to-SIP Gateway Calls via a Proxy Server
 - The Called User is Busy, page 3-14
 - A Client or Server Error Has Occurred, page 3-16
 - A Global Error Has Occurred, page 3-18
- SIP Gateway-to-SIP IP Phone
 - The Called User is Busy, page 3-21
 - The Called User Does Not Answer, page 3-22
 - A Client, Server, or Global Error Has Occurred, page 3-24

SIP Gateway-to-SIP Gateway Calls

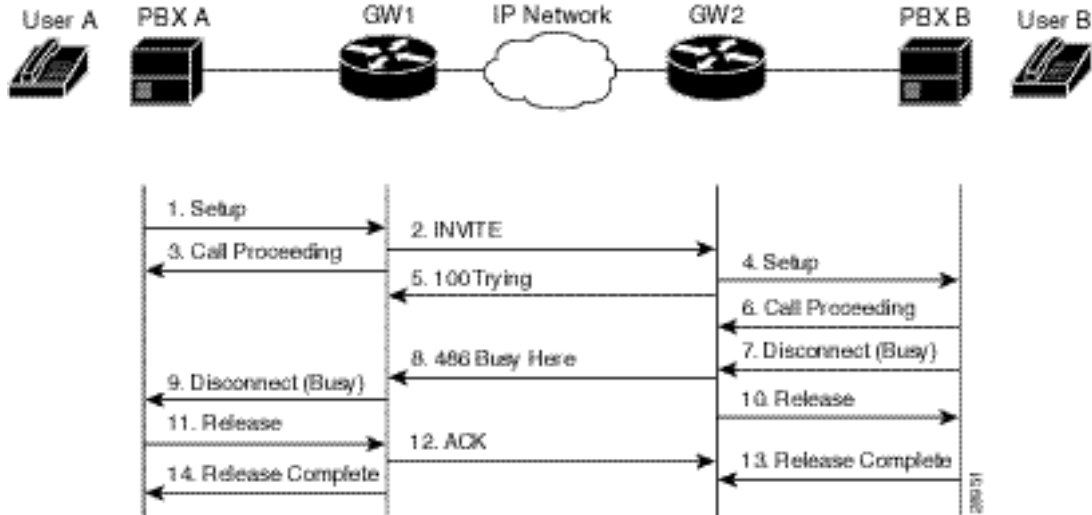
This section describes the call flows for failed SIP gateway-to-SIP gateway calls. In the following call flows, the network configuration is the same as the network configuration outlined in the “SIP Gateway-to-SIP Gateway Calls” section on page 3-1. However, instead of successfully establishing a call session, one of the following situations occurs:

- The Called User is Busy, page 3-2
- The Called User Does Not Answer, page 3-3
- A Client, Server, or Global Error Has Occurred, page 3-5

The Called User is Busy

Figure1 illustrates an unsuccessful call in which User A initiates a call to User B but User B is on the phone and is unable or unwilling to accept another call.

Figure 1 SIP Gateway-to-SIP Gateway Call—Called User is Busy



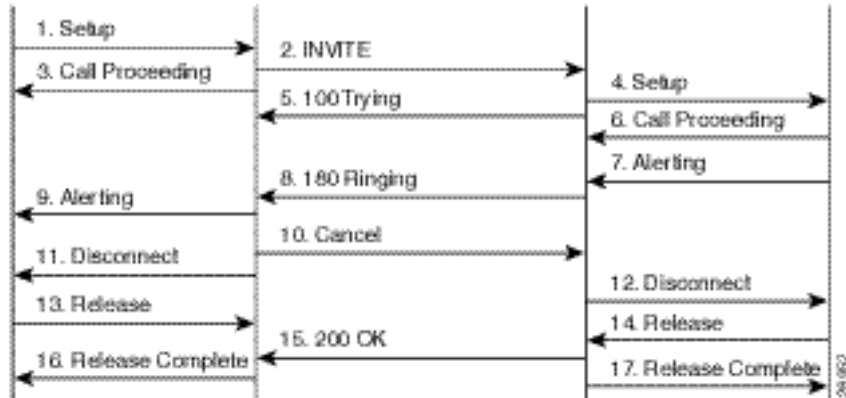
Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates a Call Setup with User B via PBX B.

Step	Action	Description
5	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying message indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
6	Call Proceeding—PBX B to SIP Gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
7	Disconnect (Busy)—PBX B to SIP Gateway 2	PBX B sends a Disconnect message to SIP gateway 2. In the Disconnect message, the cause code indicates that User B is busy. The Disconnect message starts the call session termination process.
8	486 Busy Here—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 maps the Release message cause code (Busy) to the 486 Busy Here response and sends the response to SIP gateway 1. The 486 Busy Here response is a client error response that indicates that User B's phone was successfully contacted but User B was not willing or was unable to take another call.
9	Disconnect (Busy)—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release message to PBX A. User A hears a busy tone.
10	Release—SIP Gateway 2 to PBX B	SIP gateway 2 sends a Release message to PBX B.
11	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
12	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that the 486 Busy Here response has been received.
13	Release Complete—PBX B to SIP Gateway 2	PBX B sends a Release Complete message to SIP gateway 2.
14	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.

The Called User Does Not Answer

Figure2 illustrates an unsuccessful call in which User A initiates a call to User B but User B does not answer.

Figure2 SIP Gateway-to-SIP Gateway Call—Called User Does Not Answer



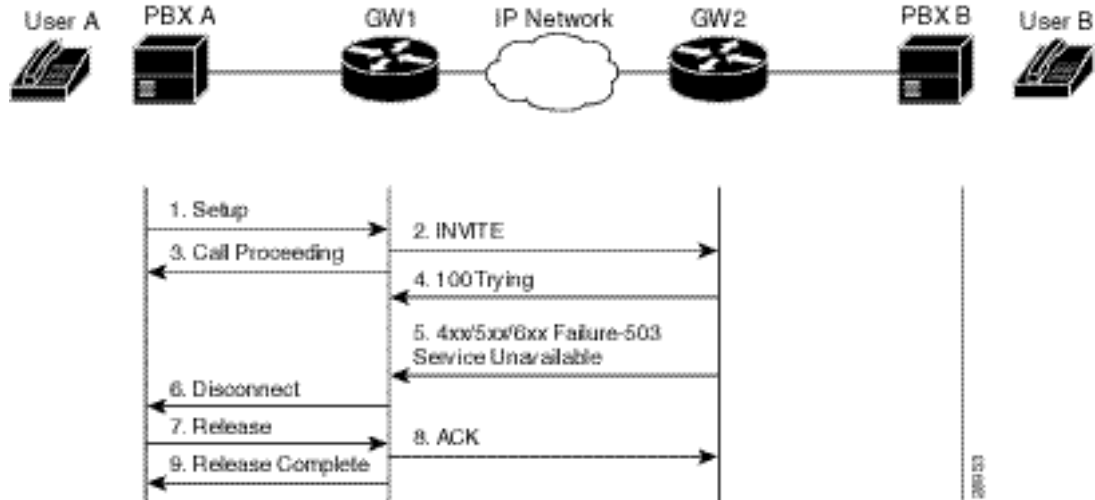
Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates a Call Setup with User B via PBX B.
5	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.

Step	Action	Description
6	Call Proceeding—PBX B to SIP Gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
7	Alerting—PBX B to SIP Gateway 2	PBX B sends an Alert message to SIP gateway 2. The message indicates that the PBX is attempting to alert the user of the call (that is to say, the phone is ringing).
8	180 Ringing—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that SIP gateway 2 has located, and is trying to alert, User B.
9	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to PBX A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response from SIP gateway 2. User A hears the ringback tone that indicates that User B is being alerted.
10	Cancel (Ring Timeout)—SIP Gateway 1 to SIP Gateway 2	Because SIP gateway 2 did not return an appropriate response within the time specified by the Expires header in the INVITE request, SIP gateway 1 sends a SIP CANCEL request to SIP gateway 2. A CANCEL request cancels a pending request with the same Call-ID, To, From, and CSeq header field values.
11	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
12	Disconnect—SIP Gateway 2 to PBX B	SIP gateway 2 sends a Disconnect message to PBX B.
13	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
14	Release—PBX B to SIP Gateway 2	PBX B sends a Release message to SIP gateway 2.
15	200 OK—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response confirms that the Cancel request has been received.
16	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A.
17	Release Complete—SIP Gateway 2 to PBX B	SIP gateway 2 sends a Release Complete message to PBX B and the call session attempt is terminated.

A Client, Server, or Global Error Has Occurred

Figure3 illustrates an unsuccessful call in which User A initiates a call to User B but there are no more channels available on the gateway. Therefore, SIP gateway 2 refuses the connection.

Figure3 SIP Gateway-to-SIP Gateway Call—Client, Server, or Global Error



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP Gateway 2	<p>SIP gateway 1 sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying message indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.

Step	Action	Description
5	Class 4xx/5xx/6xx Failure—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 determines that it does not have any more channels available, refuses the connection, and sends a SIP 503 Service Unavailable response to SIP gateway 1.</p> <p>The 503 Service Unavailable response is a class 4xx, 5xx, or class 6xx failure response. Depending on which class the failure response is, the call actions differ.</p> <p>If SIP gateway 2 sends a class 4xx failure response (a definite failure response that is a client error), the request will not be retried without modification.</p> <p>If SIP gateway 2 sends a class 5xx failure response (an indefinite failure that is a server error), the request is not terminated but rather other possible locations are tried.</p> <p>If SIP gateway 2 sends a class 6xx failure response (a global error), the search for User B is terminated because the 6xx response indicates that a server has definite information about User B, but not for the particular instance indicated in the Request-URI field. Therefore, all further searches for this user will fail.</p> <p>The call failure on SIP gateway 2 might occur before a proceeding indication from PBX B. In that case a SIP failure response is sent before the 100 Trying response.</p>
6	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
7	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
8	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that the 503 Service Unavailable response has been received.
9	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.

SIP Gateway-to-SIP Gateway Calls via a SIP Redirect Server

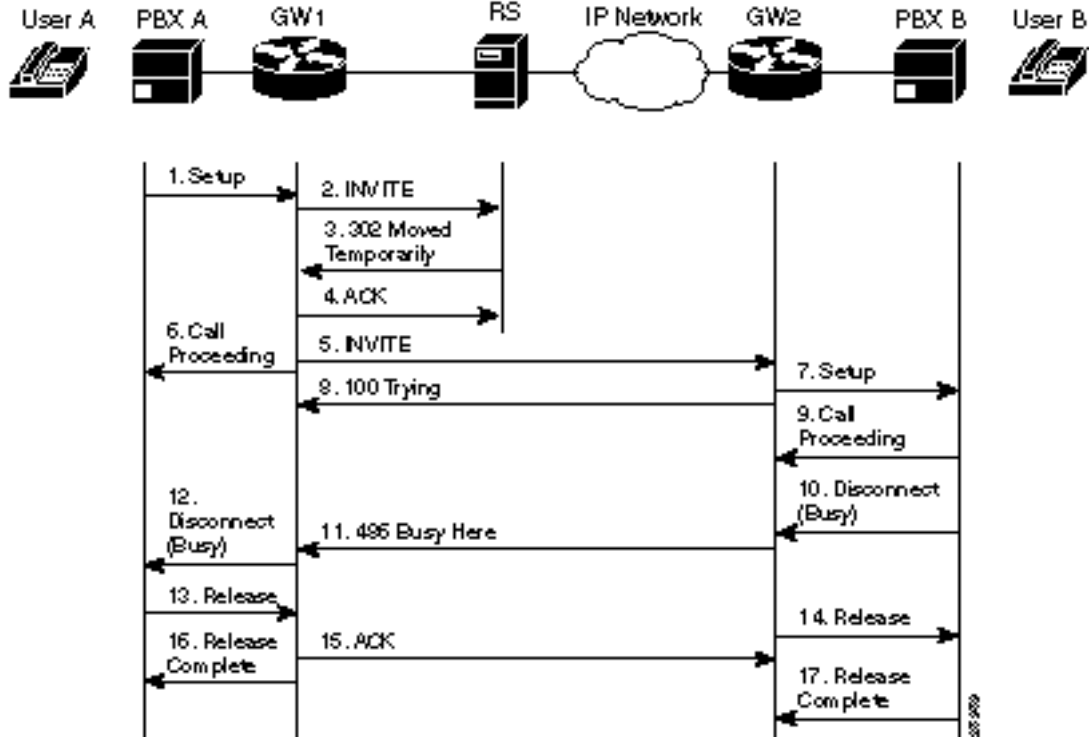
This section describes the call flows for gateway-to-gateway calls via a redirect server that have failed. In the following call flows, the network configuration is the same as the network configuration outlined in the “SIP Gateway-to-SIP Gateway Calls via a SIP Redirect Server” section on page3-7. However, instead of successfully establishing a call session, one of the following situations occurs:

- The Called User is Busy, page 3-7
- The Called User Does Not Answer, page 3-9
- A Client, Server, or Global Error Has Occurred, page 3-12

The Called User is Busy

Figure4 illustrates an unsuccessful call in which User A initiates a call to User B but User B is on the phone and is unable or unwilling to accept another call.

Figure4 SIP Gateway-to-SIP Gateway Call via a SIP Redirect Server—Called User is Busy



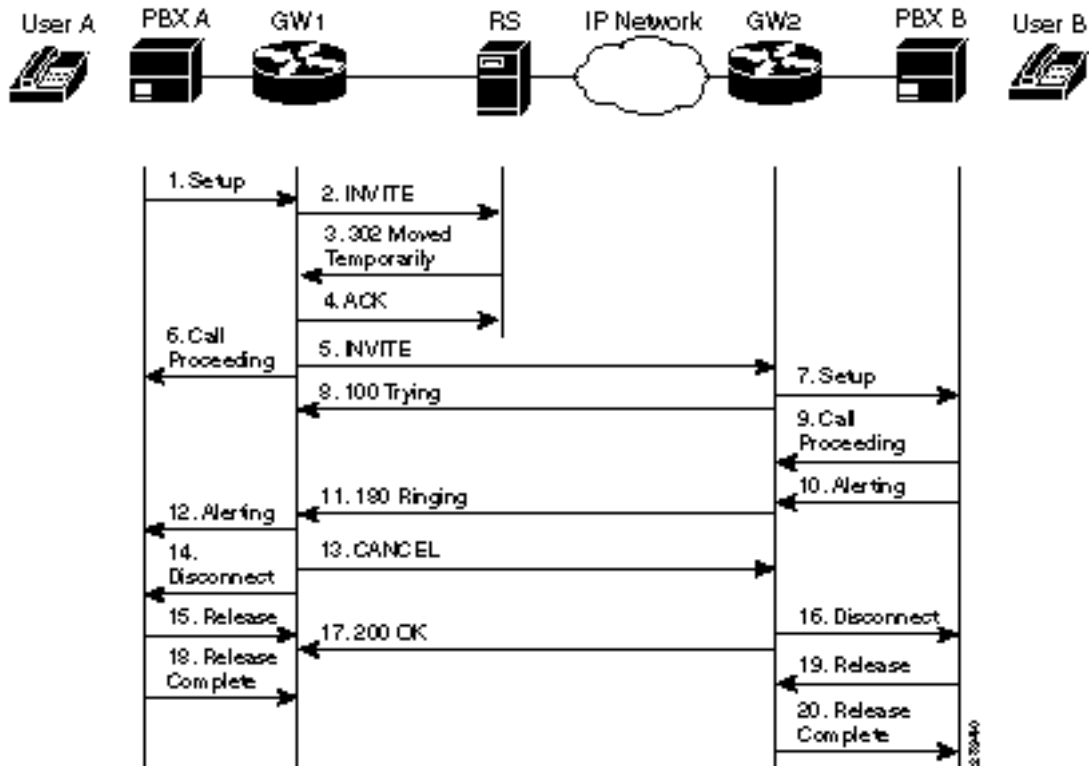
Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP Redirect Server	SIP gateway 1 sends an INVITE request to the SIP redirect server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	302 Moved Temporarily—SIP Redirect Server to SIP Gateway 1	SIP redirect server sends a 302 Moved Temporarily message to SIP gateway 1. The message indicates that User B is not available and includes instructions to locate User B.
4	ACK—SIP Gateway 1 to SIP Redirect Server	SIP gateway 1 acknowledges the 302 Moved Temporarily response with an ACK.

Step	Action	Description
5	INVITE—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a new INVITE request to User B. The new INVITE request includes the first contact listed in the 300 Multiple Choice response as the new address for User B, a higher transaction number in the CSeq field, and the same Call-ID as the first INVITE request.
6	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
7	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates a Call Setup with User B via PBXB.
8	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
9	Call Proceeding—PBX B to SIP Gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
10	Disconnect (Busy)—PBX B to SIP Gateway 2	PBX B sends a Disconnect message to SIP gateway 2. In the Disconnect message, the cause code indicates that User B is busy. The Disconnect message starts the call session termination process.
11	486 Busy Here—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 maps the Release message cause code (Busy) to the 486 Busy response and sends the response to SIP gateway 1. The 486 Busy Here response is a client error response that indicates that User B's phone was successfully contacted but User B was not willing or was unable to take another call.
12	Disconnect (Busy) —SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A. User A hears a busy tone.
13	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
14	Release—SIP Gateway 2 to PBX B	SIP gateway 1 sends a Release message to PBX B.
15	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that the 486 Busy Here response has been received.
16	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.
17	Release Complete—PBX B to SIP Gateway 2	PBX B sends a Release Complete message to SIP gateway 2.

The Called User Does Not Answer

Figure5 illustrates an unsuccessful call in which User A initiates a call to User B but User B does not answer.

Figure5 SIP Gateway-to-SIP Gateway Call via a SIP Redirect Server—Called User is Does Not Answer



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP Redirect Server	<p>SIP gateway 1 sends an INVITE request to the SIP redirect server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.

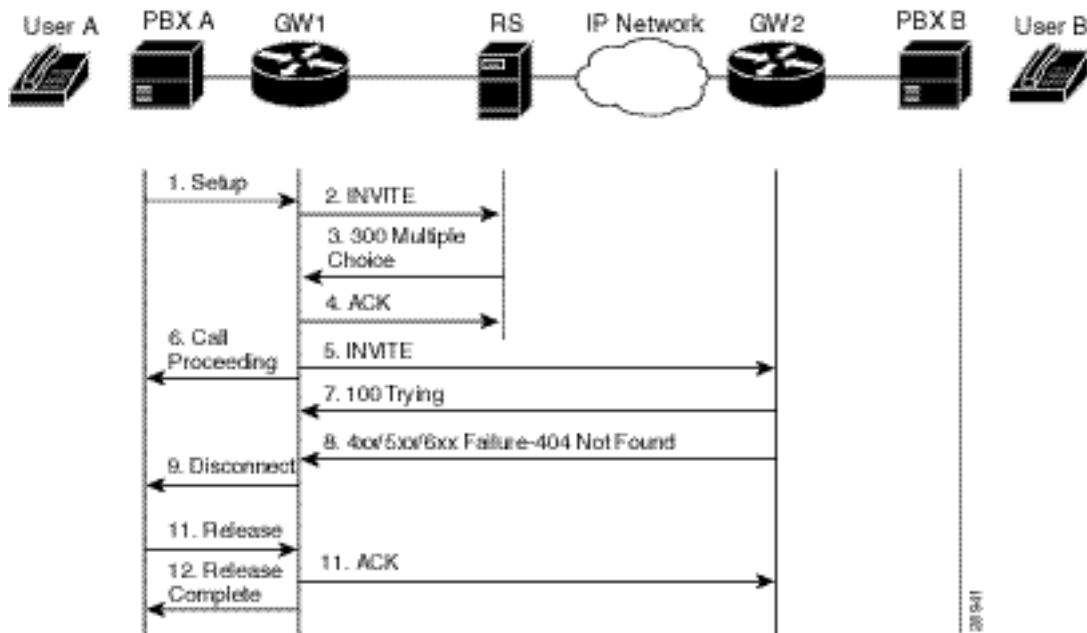
Step	Action	Description
3	302 Moved Temporarily—SIP Redirect Server to SIP Gateway 1	SIP redirect server sends a 302 Moved Temporarily message to SIP gateway 1. The message indicates that User B is not available and includes instructions to locate User B.
4	ACK—SIP Gateway 1 to SIP Redirect Server	SIP gateway 1 acknowledges the 302 Moved Temporarily response with an ACK.
5	INVITE—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a new INVITE request to User B. The new INVITE request includes a new address for User B, a higher transaction number in the CSeq field, but the same Call-ID as the first INVITE request.
6	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
7	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates a Call Setup with User B via PBXB.
8	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying message indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
9	Call Proceeding—PBX B to SIP Gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
10	Alerting—PBX B to SIP Gateway 2	PBX B sends an Alert message to SIP gateway 2. User B's phone begins to ring.
11	180 Ringing—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that SIP gateway 2 has located, and is trying to alert, User B.
12	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to PBX A.
13	CANCEL (Ring Timeout)—SIP Gateway 1 to SIP Gateway 2	Because SIP gateway 2 did not return an appropriate response within the time specified by the Expires header in the INVITE request, SIP gateway 1 sends a SIP CANCEL request to SIP gateway 2. A CANCEL request cancels a pending request with the same Call-ID, To, From, and CSeq header field values.
14	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
15	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
16	Disconnect—SIP Gateway 2 to PBX B	SIP gateway 2 sends a Disconnect message to PBX B.
17	200 OK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response confirms that the CANCEL request has been received.
18	Release Complete—PBX A to SIP Gateway 1	PBX A sends a Release Complete message to SIP gateway 1 and the call session attempt is terminated.
19	Release—PBX B to SIP Gateway 2	PBX B sends a Release message to SIP gateway 2.

Step	Action	Description
20	Release Complete—SIP Gateway 2 to PBX B	SIP gateway 2 sends a Release Complete message to PBX B.

A Client, Server, or Global Error Has Occurred

Figure6 illustrates an unsuccessful call in which User A initiates a call to User B but SIP gateway2 determines that User B does not exist at the domain specified in the INVITE request sent by SIP gateway1. SIP gateway 2 refuses the connection.

Figure 6 SIP Gateway-to-SIP Gateway Call via a SIP Redirect Server—Client, Server, or Global



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—SIP Gateway 1 to SIP Redirect Server	<p>SIP gateway 1 sends an INVITE request to the SIP redirect server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. • PBX A is identified as the initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	300 Multiple Choice—SIP Redirect Server to SIP Gateway 1	The SIP redirect server sends a 300 Multiple Choice response to SIP gateway 1. The 300 Multiple Choice response indicates that the SIP redirect server accepted the INVITE request, contacted a location server with all or part of User B's SIP URL, and the location server provided a list of alternative locations where User B might be located. The SIP redirect server returns these possible addresses to User A in the 300 Multiple Choice response.
4	ACK—SIP Gateway 1 to SIP Redirect Server	SIP gateway 1 acknowledges the 300 Multiple Choice response with an ACK.
5	INVITE—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a new INVITE request to User B. The new INVITE request includes a new address for User B, a higher transaction number in the CSeq field, but the same Call-ID as the first INVITE request.
6	Call Proceeding—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
7	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying message indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.

Step	Action	Description
8	Class 4xx/5xx/6xx Failure—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 determines that User B does not exist at the domain specified in the INVITE request sent by SIP gateway 1. SIP gateway 2 refuses the connection and sends a 404 Not Found response to SIP gateway 1.</p> <p>The 404 Not Found response is a class 4xx failure response. Depending on which class the failure response is, the call actions differ.</p> <p>If SIP gateway 2 sends a class 4xx failure response (a definite failure response that is a client error), the request will not be retried without modification.</p> <p>If SIP gateway 2 sends a class 5xx failure response (an indefinite failure that is a server error), the request is not terminated but rather other possible locations are tried.</p> <p>If SIP gateway 2 sends a class 6xx failure response (a global error), the search for User B is terminated because the 6xx response indicates that a server has definite information about User B, but not for the particular instance indicated in the Request-URI field. Therefore, all further searches for this user will fail.</p>
9	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
10	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
11	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that the 404 Not Found failure response has been received.
12	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.

SIP Gateway-to-SIP Gateway Calls via a Proxy Server

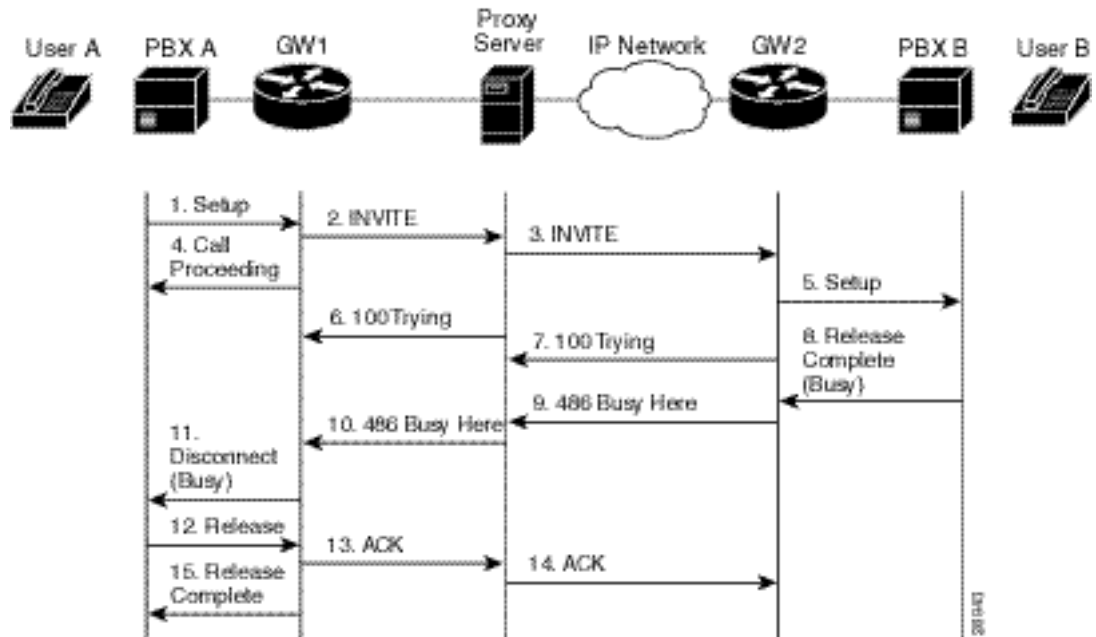
This section describes the call flows for gateway-to-gateway calls via a proxy server that have failed. In the following call flows, the network configuration is the same as the network configuration outlined in the “SIP Gateway-to-SIP Gateway Calls via a Proxy Server” section on page 3-14. However, instead of successfully establishing a call session, one of the following situations occurs:

- The Called User is Busy, page 3-14
- A Client or Server Error Has Occurred, page 3-16
- A Global Error Has Occurred, page 3-18

The Called User is Busy

Figure 7 illustrates an unsuccessful call in which User A initiates a call to User B but User B is on the phone and is unwilling or unable to accept another call.

Figure 7 SIP Gateway-to-SIP Gateway Call via a SIP Proxy Server—Called User is Busy



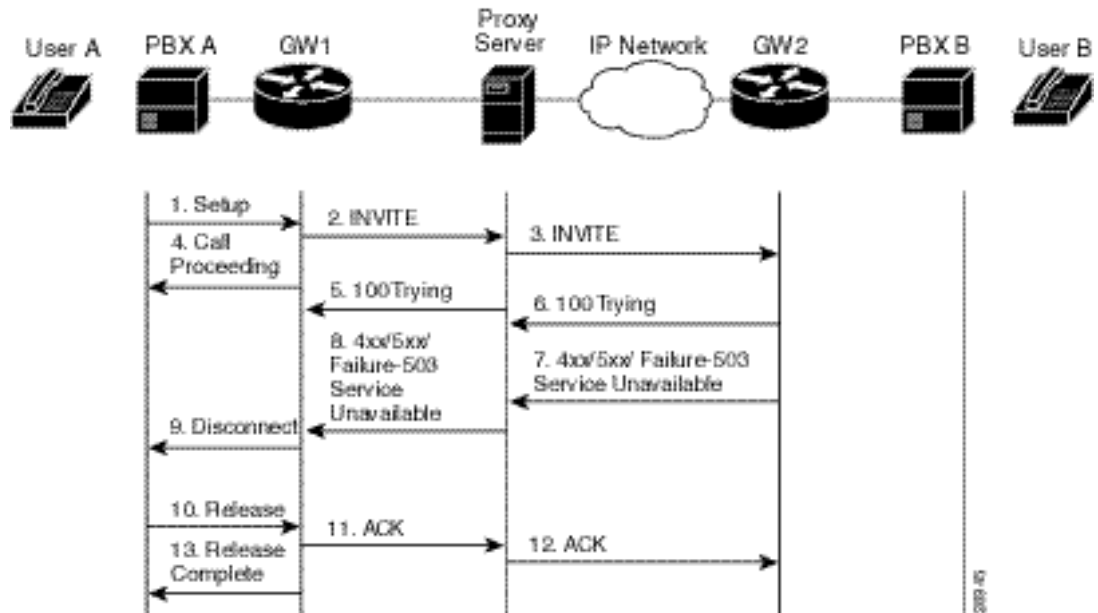
Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP Proxy Server	SIP gateway 1 sends an INVITE request to the SIP proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	INVITE—SIP Proxy Server to SIP Gateway 2	The SIP proxy server checks whether it's own address is contained in the Via field (to prevent loops), directly copies the To, From, Call-ID, and Contact fields from the request it received from SIP gateway 1, changes the Request-URI to indicate the server to which it intends to send the INVITE request, and then sends a new INVITE request to SIP gateway 2.
4	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.

Step	Action	Description
5	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from the SIP proxy server and initiates a Call Setup with UserB via PBX B.
6	100 Trying—SIP Proxy Server to SIP Gateway 1	The SIP proxy server sends a 100 Trying response to SIP gateway 1.
7	100 Trying—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 sends a 100 Trying response to the SIP proxy server.
8	Release Complete (Busy)—PBXB to SIP Gateway 2	PBX B sends a Release Complete message to SIP gateway 2. In the Release Complete message, the cause code indicates that User B is busy. The Release Complete message starts the call session termination process.
9	486 Busy Here—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 maps the Release message cause code (Busy) to the 486 Busy response and sends the response to the SIP proxy server. The 486 Busy Here response is a client error response that indicates that User B's phone was successfully contacted but User B was not willing or was unable to take another call.
10	486 Busy Here—SIP Proxy Server to SIP Gateway 1	The SIP proxy server forwards the 486 Busy response to SIP gateway 1.
11	Disconnect (Busy)—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
12	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
13	ACK—SIP Gateway 1 to SIP Proxy Server	SIP gateway 1 sends an SIP ACK to the SIP proxy server.
14	ACK—SIP Proxy Server to SIP Gateway 2	The SIP proxy server forwards the SIP ACK to SIP gateway 2. The ACK confirms that the 486 Busy Here response has been received.
15	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.

A Client or Server Error Has Occurred

Figure8 illustrates an unsuccessful call in which User A initiates a call to User B but there are no more channels available on SIP gateway 2. Therefore, SIP gateway 2 refuses the connection.

Figure 8 SIP Gateway-to-SIP Gateway Call via a SIP Proxy Server—Client or Server Error



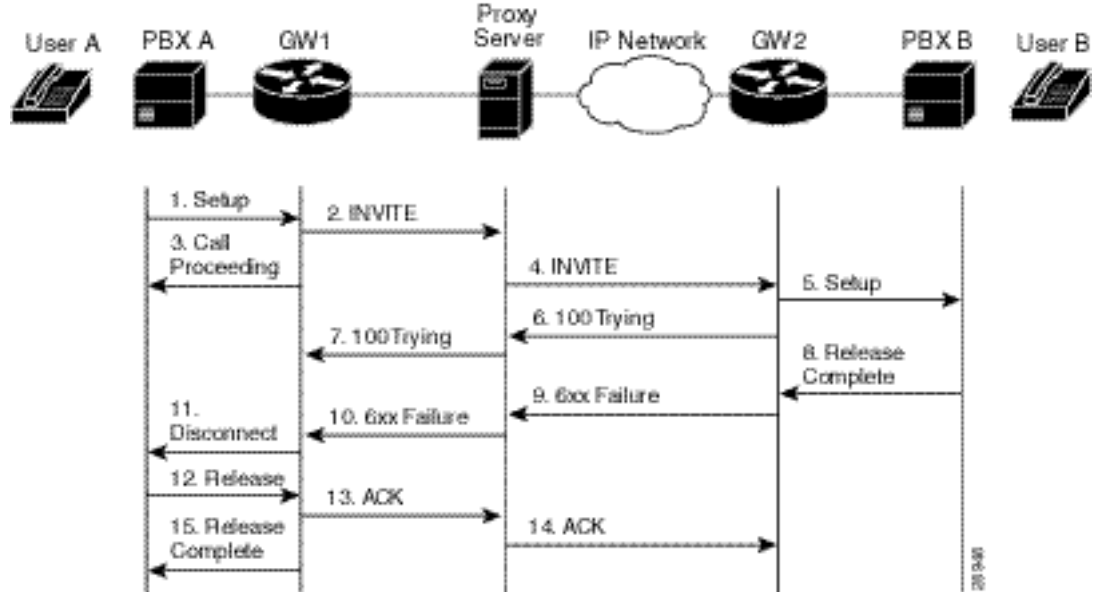
Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP Gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP Proxy Server	SIP gateway 1 sends an INVITE request to the SIP proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. • PBX A is identified as the initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	INVITE—SIP Proxy Server to SIP Gateway 2	The SIP proxy server checks whether it's own address is contained in the Via field (to prevent loops), directly copies the To, From, Call-ID, and Contact fields from the request it received from SIP gateway 1, changes the Request-URI to indicate the server to which it intends to send the INVITE request, and then sends a new INVITE request to SIP gateway 2.
4	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.

Step	Action	Description
5	100 Trying—SIP Proxy Server to SIP Gateway 1	The SIP proxy server sends a 100 Trying response to SIP gateway 1.
6	100 Trying—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 sends a 100 Trying response to the SIP proxy server.
7	Class 4xx/5xx/6xx Failure—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 determines that it does not have any more channels available, refuses the connection, and sends a SIP 503 Service Unavailable response to the SIP proxy server.
8	Class 4xx/5xx/6xx Failure—SIP Proxy Server to SIP Gateway 1	The SIP proxy server forwards the SIP 503 Service Unavailable response to SIP gateway 1.
9	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
10	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
11	ACK—SIP Gateway 1 to SIP Proxy Server	SIP Gateway 1 sends an ACK to the SIP proxy server.
12	ACK—SIP Proxy Server to SIP Gateway 2	The SIP proxy server forwards the SIP ACK to SIP Gateway 2. The ACK confirms that the 503 Service Unavailable response has been received.
13	Release Complete—SIP Gateway 1 to PBX A	SIP Gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.

A Global Error Has Occurred

Figure9 illustrates an unsuccessful call in which User A initiates a call to User B and receives a class 6xx response.

Figure9 SIP Gateway-to-SIP Gateway Call via a SIP Proxy Server—Global Error Response



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP Proxy Server	SIP gateway 1 sends an INVITE request to the SIP proxy server. The INVITE request is an invitation to User B to participate in a call session. In the INVITE request: <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	INVITE—SIP Proxy Server to SIP Gateway 2	The SIP proxy server checks whether it's own address is contained in the Via field (to prevent loops), directly copies the To, From, Call-ID, and Contact fields from the request it received from SIP gateway 1, changes the Request-URI to indicate the server to which it intends to send the INVITE request, and then sends a new INVITE request to SIP gateway 2.

Step	Action	Description
5	Setup—SIP Gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from the SIP proxy server and initiates a Call Setup with UserB via PBX B.
6	100 Trying—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 sends a 100 Trying response to the SIP proxy server. The SIP proxy server might or might not forward the 100 Trying response to SIP gateway 1.
7	100 Trying—SIP Proxy Server to SIP Gateway 1	The SIP proxy server forwards the 100 Trying response to SIP gateway 1.
8	Release Complete—PBX B to SIP Gateway 2	PBX B sends a Release Complete message to SIP gateway 2. The Release Complete message starts the call session termination process.
9	6xx Failure—SIP Gateway 2 to SIP Proxy Server	SIP gateway 2 sends a class 6xx failure response (a global error) to the SIP proxy server. A class 6xx failure response indicates that a server has definite information about User B, but not for the particular instance indicated in the Request-URI field. All further searches for this user will fail, therefore the search is terminated. The SIP proxy server must forward all class 6xx failure responses to the client.
10	6xx Failure—SIP Proxy Server to SIP Gateway 1	The SIP proxy server forwards the 6xx failure to SIP gateway 1.
11	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
12	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
13	ACK—SIP Gateway 1 to SIP Proxy Server	SIP gateway 1 sends an ACK to the SIP proxy server.
14	ACK—SIP Proxy Server to SIP Gateway 2	The SIP proxy server sends an ACK to SIP gateway 2. The ACK confirms that the 6xx failure response has been received.
15	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.

SIP Gateway-to-SIP IP Phone

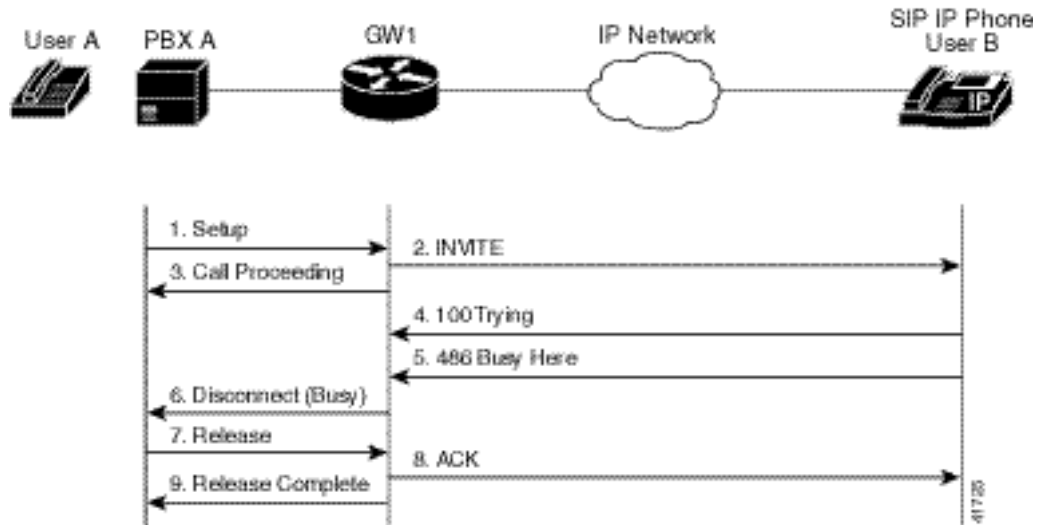
This section describes the call flows for failed SIP gateway-to-SIP IP phone calls. In the following call flows, the network configuration is the same as the network configuration outlined in the “SIP Gateway-to-SIP IP Phone” section on page 3-20. However, instead of successfully establishing a call session, one of the following situations occurs:

- The Called User is Busy, page 3-21
- The Called User Does Not Answer, page 3-22
- A Client, Server, or Global Error Has Occurred, page 3-24

The Called User is Busy

Figure10 illustrates an unsuccessful call in which User A initiates a call to User B but User B is on the phone and is unable or unwilling to take another call.

Figure10 SIP Gateway-to-SIP IP Phone—Called User is Busy



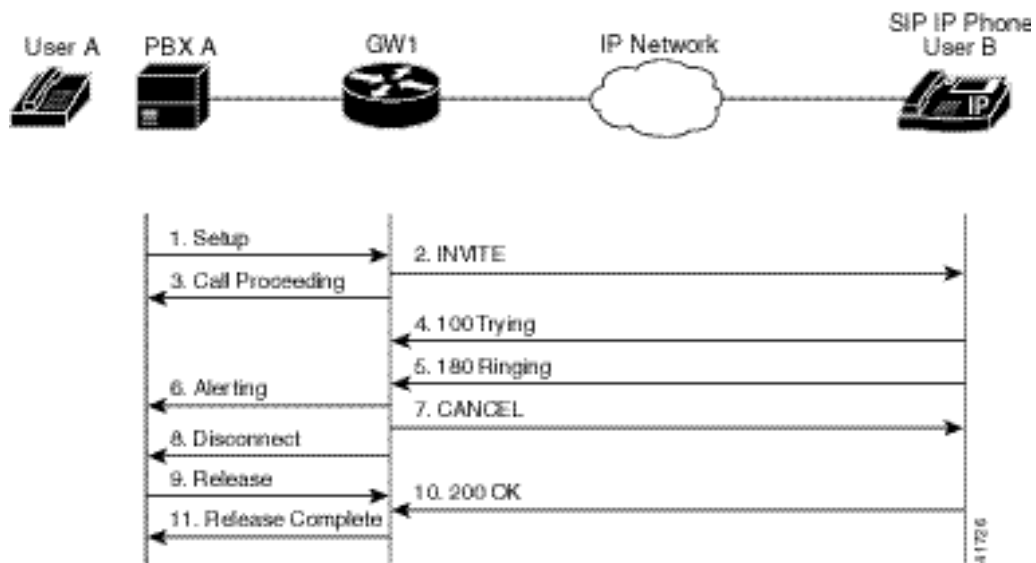
Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 maps the SIP URL phone number to a dial-peer. The dial-peer includes the IP address and the port number of the SIP enabled entity to contact. SIP gateway 1 sends an INVITE request to the address it receives in the dial peer which, in this scenario, is the SIP IP phone. In the INVITE request: <ul style="list-style-type: none"> • The IP address of the SIP IP phone is inserted in the Request-URI field. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which the SIP gateway is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.

Step	Action	Description
4	100 Trying—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 100 Trying response to SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by the SIP IP phone.
5	486 Busy Here—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 486 Busy Here response to SIP gateway 1. The 486 Busy Here response is a client error response that indicates that User B was successfully contacted but User B was not willing or was unable to take the call.
6	Disconnect (Busy)—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
7	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
8	ACK—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 sends an ACK to the SIP IP phone. The ACK confirms that User A has received the 486 Busy Here response. The call session attempt is now being terminated.
9	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.

The Called User Does Not Answer

Figure11 illustrates the call flow in which User A initiates a call to User B but User B does not answer.

Figure11 SIP Gateway-to-SIP IP Phone—Called User Does Not Answer



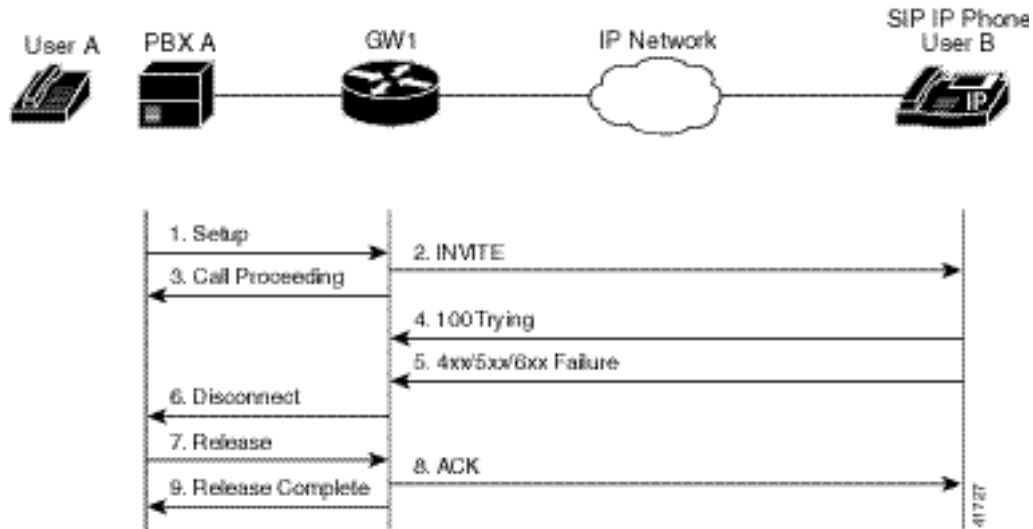
Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—SIP Gateway 1 to SIP IP Phone	<p>SIP gateway 1 maps the SIP URL phone number to a dial-peer. The dial-peer includes the IP address and the port number of the SIP enabled entity to contact. SIP gateway 1 sends an INVITE request to the address it receives in the dial peer which, in this scenario, is the SIP IP phone.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The IP address of the SIP IP phone is inserted in the Request-URI field. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which the SIP gateway is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	100 Trying—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 100 Trying response to SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by the SIP IP phone.
5	180 Ringing—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 180 Ringing response to SIP gateway 1. The 180Ringing response indicates that the user is being alerted.
6	Alerting—SIP Gateway 1 to PBX A	SIP gateway 1 sends an Alert message to PBX A. The Alert message indicates that SIP gateway 1 has received a 180 Ringing response from the SIP IP phone. User A hears the ringback tone that indicates that User B is being alerted.
7	CANCEL (Ring Timeout)—SIP Gateway 1 to SIP IP Phone	Because SIP gateway 1 did not return an appropriate response within the time specified by the Expires header in the INVITE request, SIP gateway 1 sends a SIP CANCEL request to SIP gateway 2. A CANCEL request cancels a pending request with the same Call-ID, To, From, and CSeq header field values.
8	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Disconnect message to PBX A.
9	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.
10	200 OK—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 200 OK response to SIP gateway 1. The 200 OK response confirms that User A has received the Cancel request. The call session attempt is now being terminated.
11	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session is terminated.

A Client, Server, or Global Error Has Occurred

Figure12 illustrates an unsuccessful call in which User A initiates a call to User B and receives a class 4xx, 5xx, or 6xx response.

Figure12 SIP Gateway-to-SIP IP Phone—Client, Server, or Global Error



Step	Action	Description
1	Setup—PBX A to SIP Gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 maps the SIP URL phone number to a dial-peer. The dial-peer includes the IP address and the port number of the SIP enabled entity to contact. SIP gateway 1 sends an INVITE request to the address it receives in the dial peer which, in this scenario, is the SIP IP phone. In the INVITE request: <ul style="list-style-type: none"> • The IP address of the SIP IP phone is inserted in the Request-URI field. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified. • The port on which the SIP gateway is prepared to receive the RTP data is specified.
3	Call Proceeding—SIP Gateway1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.

Step	Action	Description
4	100 Trying—SIP IP Phone to SIP Gateway 1	The SIP IP phone sends a 100 Trying response to SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by the SIP IP phone.
5	4xx/5xx/6xx Failure—SIP IP Phone to SIP Gateway 1	<p>The SIP IP phone sends a class 4xx, 5xx, or class 6xx failure response to SIP gateway1. Depending on which class the failure response is, the call actions differ.</p> <p>If the SIP IP phone sends a class 4xx failure response (a definite failure response that is a client error), the request will not be retried without modification.</p> <p>If the SIP IP phone sends a class 5xx failure response (an indefinite failure that is a server error), the request is not terminated but rather other possible locations are tried.</p> <p>If the SIP IP phone sends a class 6xx failure response (a global error), the search for User B is terminated because the 6xx response indicates that a server has definite information about User B, but not for the particular instance indicated in the Request-URI field. Therefore, all further searches for this user will fail.</p>
6	Disconnect—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release message to PBX A.
7	Release—PBX A to SIP Gateway 1	PBX A sends a Release message to SIP gateway 1.
8	ACK—SIP Gateway 1 to SIP IP Phone	SIP gateway 1 sends an ACK to the SIP IP phone. The ACK confirms that User A has received the 4xx/5xx/6xx failure response. The call session attempt is now being terminated.
9	Release Complete—SIP Gateway 1 to PBX A	SIP gateway 1 sends a Release Complete message to PBX A and the call session attempt is terminated.



Cisco SIP Compliance Reference Information

This section describes how the Cisco SIP User Agent and the Cisco SIP Gateway comply with the IETF definition of SIP as described in RFC 2543.

This section contains compliance information on the following:

- SIP Functions, page 4-2
- SIP Methods, page 4-2
- SIP Responses, page 4-3
- SIP Header Fields, page 4-11
- SIP Transport Layer Protocols, page 4-13
- SIP Security, page 4-13
- SIP Session Description Protocol (SDP) Usage, page 4-13
- SIP DNS Records Usage, page 4-14
- SIP DTMF Relay, page 4-14
- Session Timer Draft, page 4-14
- Fax Relay, page 4-15
- SIP T.37/T.38 Fax Relay, page 4-15
- SIP synchronization with RSVP for QoS, page 4-17

SIP Functions

Function	Supported?	Comments
User Agent Client (UAC)	Yes	
User Agent Server (UAS)	Yes	
Proxy Server/Redirect Server/Registrar	No.	The SIP gateway does not have the proxy or redirect server functionality, but can work with an external proxy or redirect server.

SIP Methods

There are six methods used by the SIP gateway.

Method	Supported?	Comments
INVITE	Yes	The gateway supports mid-call INVITES with the same call ID but different SDP session parameters (to change the transport address).
ACK	Yes	
OPTIONS	Yes	The gateway does not generate OPTIONS. However, it will respond to OPTIONS requests.
BYE	Yes	
CANCEL	Yes	
COMET	Yes	COnditions MET. Used in QoS implementation to indicate to other endpoint whether or not the conditions have been met, i.e. resources reserved.
NOTIFY	Yes	Used in implementation of REFER to let initiator of the REFER know the outcome of the transfer.
PRACK	Yes	PRovisional ACKnowledgement. Used in reliable provisional responses.
REFER	Yes	Gateways will respond to a REFER, but they do not generate a REFER for transfer.
REGISTER	No	
SUBSCRIBE	Yes	Gateway processes SUBSCRIBE for selected applications such as telephony events (DTMF) and for Message Waiting Indication (MWI).
INFO	Yes	Gateway does not generate INFO method. It can handle incoming INFO methods

SIP Responses

Cisco IOS Release 12.2(10)T supports the following SIP Responses:

- 1xxResponse—Information Responses, page 4-4
- 2xxResponse—Successful Responses, page 4-4
- 3xxResponse—Redirection Responses, page 4-5
- 4xxResponse—Request Failure Responses, page 4-5
- 5xxResponse—Server Failure Responses, page 4-10
- 6xxResponse—Global Responses, page 4-11

1xx Response—Information Responses

1xx Response	Comments
<p>100 Trying</p> <p>This response indicates that action is being taken on behalf of the caller, but that the callee has not yet been located.</p>	<p>The SIP gateway generates this response for an incoming INVITE.</p> <p>Upon receiving this response, the gateway stops retransmitting INVITEs. It then waits for a 180 Ringing or 200 OK response.</p>
<p>180 Ringing</p> <p>This response indicates that the callee has been located and is being notified of the call.</p>	<p>The SIP gateway generates a 180 Ringing response when the called party has been located and is being alerted.</p> <p>Upon receiving this response, the gateway waits for a 200 OK response.</p>
<p>181 Call is being forwarded</p> <p>This response indicates that the call is being rerouted to another destination.</p>	<p>The SIP gateway does not generate this response.</p>
<p>182 Queued</p> <p>This response indicates that the callee is not currently available but that they have elected to queue the call rather than reject it.</p>	<p>Upon receiving this response, the gateway processes the responses the same way that it processes a 100 Trying response.</p>
<p>183 Session progress</p> <p>This response is used to perform inband alerting for the caller.</p>	<p>The SIP gateway generates a 183 Session progress response when it receives an ISDN Progress message with an appropriate media indication from PSTN.</p>

2xx Response—Successful Responses

2xx Response	Comments
<p>200 OK</p> <p>This response indicates that the request has been successfully processed. The action taken depends on the request made.</p>	<p>The SIP gateway generates this response when the PBX indicates that the user has answered the phone.</p> <p>Upon receiving this response, the gateway forwards the response to the corresponding party and responds with an ACK.</p>

3xx Response—Redirection Responses

3xx Response	Comments
<p>300 Multiple choices</p> <p>This response indicates that the address resolved to more than one location. All locations are provided and the user or UA is allowed to select which location to use.</p>	<p>The SIP gateway does not generate this response. Upon receiving this response, the gateway contacts the new address in the Contact header field.</p>
<p>301 Moved permanently</p> <p>This response indicates that the user is no longer available at the specified location. An alternate location is included in the header.</p>	
<p>302 Moved temporarily</p> <p>This response indicates that the user is temporarily unavailable at the specified location. An alternate location is included in the header.</p>	
<p>305 Use proxy</p> <p>This response indicates that the caller must use a proxy to contact the callee.</p>	
<p>380 Alternative service</p> <p>This response indicates that the call was unsuccessful, but that alternative services are available.</p>	

4xx Response—Request Failure Responses

4xx Response	Comments
<p>400 Bad Request</p> <p>This response indicates that the request could not be understood because of an illegal format.</p>	<p>The SIP gateway generates a 400 Bad Request response for a badly formed request.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>

4xx Response	Comments
<p>401 Unauthorized</p> <p>This response indicates that the request requires user authentication.</p>	
<p>402 Payment required</p> <p>This response indicates that payment is required to complete the call.</p>	<p>The SIP gateway does not generate this response.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>403 Forbidden</p> <p>This response indicates that the server has received and understood the request but will not provide the service.</p>	
<p>404 Not Found</p> <p>This response indicates that the server has definite information that the user does not exist in the specified domain.</p>	<p>The SIP gateway generates this response if it is unable to locate the callee.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>405 Method Not Allowed</p> <p>This response indicates that the method specified in the request is not allowed. The response contains a list of allowed methods.</p>	<p>The SIP gateway generates this response if an invalid method is specified in the request.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>406 Not Acceptable</p> <p>This response indicates that the requested resource is capable of generating only responses that have content characteristics not acceptable as specified in the accept header of the request.</p>	<p>The SIP gateway does not generate this response.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>

4xx Response	Comments
<p>407 Proxy authentication required</p> <p>This response is similar to the 401 Unauthorized response. However, this response indicates that the client must first authenticate itself with the proxy.</p>	<p>The SIP gateway does not generate this response.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>408 Request timeout</p> <p>This response indicates that the server could not produce a response before the Expires time out.</p>	
<p>409 Conflict</p> <p>This response indicates that the request could not be processed because of a conflict with the current state of the resource.</p>	
<p>410 Gone</p> <p>This response indicates that a resource is no longer available at the server and no forwarding address is known.</p>	<p>The SIP gateway generates this response if the PSTN returns a cause code of unallocated number.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>411 Length required</p> <p>This response indicates that the user refuses to accept the request without a defined content length.</p>	<p>The SIP gateway does not generate this response.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>413 Request entity too large</p> <p>This response indicates that server refuses to process the request because it is larger than the server is willing or able to process.</p>	
<p>414 Request-URI too long</p> <p>This response indicates that the server refuses to process the request because the Request-URI is too long for the server to interpret.</p>	
<p>415 Unsupported media</p> <p>This response indicates that the server refuses to process the request because the format of the body is not supported by the destination endpoint.</p>	

4xx Response	Comments
<p>420 Bad extension</p> <p>This response indicates that the server could not understand the protocol extension indicated in the Require header.</p>	<p>The SIP gateway generates this response if it cannot understand the service requested in the Require header field.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>480 Temporarily unavailable</p> <p>This response indicates that the callee was contacted but is temporarily unavailable.</p>	<p>The SIP gateway generates this response if the callee is unavailable (for example, the callee does not answer the phone within a certain amount of time or the called number does not exist or is no longer in service).</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>481 Call leg/transaction does not exist</p> <p>This response indicates that the server is ignoring the request because it was either a BYE for which there was no matching legID or a CANCEL for which there was no matching transaction.</p>	<p>The SIP gateway generates this response if the call leg ID or transaction cannot be identified.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>482 Loop detected</p> <p>This response indicates that the server received a request that included itself in the path.</p>	<p>The SIP gateway does not generate this response.</p>
<p>483 Too many hops</p> <p>This response indicates that the server received a request that required more hops than allowed by the Max-Forwards header.</p>	
<p>484 Address incomplete</p> <p>This response indicates that the server received a request containing an incomplete address.</p>	
<p>485 Ambiguous</p> <p>This response indicates that the server received a request in which the callee address was ambiguous. It can provide possible alternate addresses.</p>	<p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>

4xx Response	Comments
<p>486 Busy here</p> <p>This response indicates that the callee was contacted but that their system is unable to take additional calls.</p>	<p>The SIP gateway generates this response if the callee was contacted but was busy.</p> <p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>487 Request cancelled</p> <p>This response indicates that the request was terminated by a BYE or CANCEL request.</p>	<p>The SIP gateway generates this response to an unexpected BYE or CANCEL received for a request.</p>
<p>488 Not Acceptable Media</p> <p>This response indicates an error in handling the request at this time.</p>	<p>The SIP gateway generates this response if the media negotiation fails.</p>
<p>422 Session Timer Too Small</p>	<p>It is generated by the gateway (UAS) when a request contains a Session-Expires header with a duration that is below the minimum timer for the gateway server. The 422 response MUST contain a Min-SE header with a minimum timer for that server.</p>

5xx Response—Server Failure Responses

5xx Response	Comments	
<p>500 Server internal error</p> <p>This response indicates that the server or gateway encountered an unexpected error that prevented it from processing the request.</p>		
<p>501 Not implemented</p> <p>This response indicates that the server or gateway does not support the functions required to complete the request.</p>		
<p>502 Bad gateway</p> <p>This response indicates that the server or gateway received an invalid response from a downstream server.</p>		<p>The SIP gateway generates this response if it encountered an unexpected error that prevented it from processing the request.</p>
<p>503 Service unavailable</p> <p>This response indicates that the server or gateway is unable to process the request due to an overload or maintenance problem.</p>		<p>Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>504 Gateway timeout</p> <p>This response indicates that the server or gateway did not receive a timely response from another server (such as a location server).</p>		
<p>505 Version not supported</p> <p>This response indicates that the server or gateway does not support the version of the SIP protocol used in the request.</p>		
<p>580 Precondition failed</p>		<p>The SIP gateway uses this response code to indicate a failure in having Qos preconditions met for a call.</p>

6xx Response—Global Responses

6xx Response	Comments
<p>600 Busy everywhere</p> <p>This response indicates that the callee was contacted but that the callee is busy and cannot take the call at this time.</p>	<p>The SIP gateway does not generate this response. Upon receiving this response, the gateway initiates a graceful call disconnect and clears the call.</p>
<p>603 Decline</p> <p>This response indicates that the callee was contacted but cannot or does not want to participate in the call.</p>	
<p>604 Does not exist anywhere</p> <p>This response indicates that the server has authoritative information that the callee does not exist in the network.</p>	
<p>606 Not acceptable</p> <p>This response indicates that the callee was contacted, but that some aspect of the session description was unacceptable.</p>	

SIP Header Fields

Header Field	Supported?
Accept	Yes
Accept-Encoding	No
Accept-Language	No
Allow	Yes
Also	Yes
Authorization	No
Call-ID	Yes
Contact	Yes
Content-Encoding	No
Content-Length	Yes
Content-Disposition	Yes
Content-Type	Yes
Cseq	Yes
Date	Yes

Header Field	Supported?
Diversion	Yes
Encryption	No
Event	Yes
Expires	Yes
From	Yes
Hide	No
Max-Forwards	Yes
Organization	No
Min-SE	Yes
Priority	No
Proxy-Authenticate	No
Proxy Authorization	No
Proxy-Require	No
RAcr	Yes
RSeq	Yes
Record-Route	Yes
Require	Yes
Response-Key	No
Retry-After	No
Refer-to	Yes
Referred-By	Yes
Replaces	Yes
Requested-By	Yes
Route	Yes
Server	Yes
Session-Expires	Yes
Subject	No
Supported	Yes
Timestamp	Yes
To	Yes
Unsupported	Yes
User-Agent	Yes
Via	Yes
Warning	Yes
WWW-Authenticate	No

SIP Transport Layer Protocols

Transport Layer Protocol	Supported?	Comments
Unicast UDP	Yes	None.
Multicast UDP	No	
TCP	Yes	None.

SIP Security

Encryption

Encryption Mode	Supported?	Comments
End-to-end Encryption	No	IPSEC can be used for security.
Privacy of SIP Responses	No	None.
Hop-by-Hop Encryption	No	IPSEC can be used for security.
Via Field Encryption	No	

Authentication

Encryption Mode	No
Basic Authentication	No
Digest Authentication	No
Proxy Authentication	No
PGP	No

SIP Session Description Protocol (SDP) Usage

SDP Headers	Supported?
v—Protocol version	Yes
o—Owner/creator and session identifier	Yes
s—Session name	Yes
c—Connection information	Yes
t—Time the session is active	Yes

SDP Headers	Supported?
m—Media name and transport address	Yes
a—Attribute. The primary means for extending SDP and tailoring it to particular applications or media.	Yes

SIP DNS Records Usage

DNS Resource Record Type	Supported?
Type A	Yes
Type SRV	Yes. Both RFC2052 and RFC2782 formatting are supported.

SIP DTMF Relay

Cisco gateways support DTMF Relay in accordance with RFC2833. The DTMF Relay method is defined as “rtp-nte” in gateway configuration. It is based on the transmission of Named Telephony Events (NTE) and DTMF digits over RTP stream. The default RTP payload used for transmission of DTMF is 10.

Methods	Supported?
RFC 2833 AVT Tones	Yes
Cisco RTP Method	Yes

Session Timer Draft

This supports draft-ietf-sip-session-timer-08.txt

Fields	Supported?
Supported	Yes



Note

Cisco gateways do not initiate use of keepalives for any calls that it originates or terminates. If the “far side” wants to use keepalives, the gateway complies.

Fax Relay

Methods	Supported?	Comment
T.38 fax relay	Yes	
Cisco Fax relay	Yes	Not supported on 5350 and 5400 platform

SIP T.37/T.38 Fax Relay

Cisco gateways support T.38/T.37 fax relay/store and forward mechanisms. The following table lists requirements based on Annex-D of ITU recommendation T.38. It indicates whether the requirements are supported.

Requirement	Description	Mandatory/ Optional	Supported
<SIPt38-01>	T.38 over SIP shall be implemented as described in ANNEX D to Recommendation T.38 - SIP/SDP Call Establishment Procedures (previously known as TD0262rev2)	Mandatory	Yes
<SIPt38-02>	SIP-enabled VoIP gateways must be able to detect the CNG, CED fax tones and/or the preamble flag sequence transmitted inside the audio RTP streams	Mandatory	Yes (CED, V.21 preamble detected now. Future DSP releases will detect CNG. CNG is an optional tone.)
<SIPt38-03>	Detection of a fax transmission MUST be performed by the receiving gateway by recognizing the CED tone	Mandatory	Yes
<SIPt38-04>	If CED is not present, fax transmission MUST be detected by the receiving gateway by recognizing the Preamble flag sequence	Mandatory	Yes
<SIPt38-05>	Upon detection of the fax transmission, the gateway MUST initiate switch over to T.38 fax mode by sending a re-INVITE with proper SDP description	Mandatory	Yes
<SIPt38-06>	To prevent glare, even if the transmitting gateway detects the fax transmission (CNG tone), it MUST not initiate the switch over to T.38 fax mode	Mandatory	Yes

Requirement	Description	Mandatory/ Optional	Supported
<SIPt38-07>	If a SIP session starts with audio capabilities, then switches to fax, the capability must be provided for the session to switch back to audio mode at the end of the fax transmission	Mandatory	Yes
<SIPt38-08>	Support of SIP T.38 fax calls over TCP is desired	Desirable	UDP only
<SIPt38-09>	The SDP protocol type UDPTL (Facsimile UDP transport Layer) must be supported as lately defined by IANA	Mandatory	Yes
<SIPt38-10>	Following SDP attributes as defined by IANA to support T.38 fax sessions must be incorporated: Registered SDP Protocol format , MIME media type image/t38: <ul style="list-style-type: none"> • MIME media type name: image • MIME subtype name: t38 	Mandatory	Yes
<SIPt38-11>	The following SDP attributes as defined by IANA to support T.38 sessions must be incorporated. <ol style="list-style-type: none"> 1. T38FaxVersion 2. T38maxBitRate 3. T38FaxFillBitRemoval 4. T38FaxTranscodingMMR 5. T38FaxTranscodingJBIG 6. T38FaxRateManagement 7. T38FaxMaxBuffer 8. T38FaxMaxDatagram 9. T38FaxUdpEC 	Mandatory	Yes
<SIPt38-12>	Any SIP-enabled Cisco gateway supporting T.38 must be able to interoperate with gateways from Cisco and other vendors.	Mandatory	Yes
<SIPt38-13>	Interoperability with gateways supporting T.38 over H.323 is desired and should be addressed as part of overall SIP H.323 interoperability implementation.	Optional	No

Requirement	Description	Mandatory/ Optional	Supported
<SIPt38-14>	Configuration of SIP enabled gateways should include management of SIP T.38 specific configurable choices.	Mandatory	Yes. The following are configurable: <ul style="list-style-type: none"> • bitrate • TCP/UDP (UDP only) • hs, ls redundancy • ECM
<SIPt38-15>	Tracking and reporting of SIP T.38 activity on the gateways is desired. This includes generation of Call Detail Records (CDR) for SIP T.38 fax calls.	Mandatory	Yes
<SIPt38-16>	The security mechanisms provided in RFC2543 apply. Message authentication can be performed on SIP INVITEs and BYE.	Optional	No

SIP synchronization with RSVP for QoS

This is a multi-source draft dated 7/2000.

Fields	Supported?
Supported	Yes



SIP Gateway Support for Third Party Call Control

This appendix describes the SIP Gateway Support for Third-Party Call Control feature that is introduced in Cisco IOS Release 12.1(5)XM. In addition to the SIP Gateway Support for Third-Party Call Control feature, this appendix describes the SIP gateway's ability to process Fully Qualified Domain Names (FQDNs) in the SIP media.

Third-party Call Control via Delayed Media

The SIP gateway support for third-party call control enables one endpoint (for example, a call controller) to create, modify, or terminate calls between other endpoints via *delayed media negotiation*. A delayed media negotiation is one where the Session Description Protocol (SDP) information is not completely advertised in the initial call setup.

When delayed media negotiation takes place between two SIP endpoints (User A and User B) and a controlling endpoint (call controller) that is creating the call between User A and User B, the following actions take place:

1. The call controller sends an INVITE request to the first endpoint to be included in the call (User A). The INVITE does not contain an SDP body.
2. The call controller sends an INVITE request to the second endpoint to be included in the call (User B). The INVITE request does not contain an SDP body.
3. User A sends a 200 OK response to the call controller. The 200 OK response contains User A's SDP information.
4. User B sends a 200 OK response to the call controller. The 200 OK response contains User B's SDP information.
5. On receiving a 200 OK response from User A, the call controller sends an ACK response to User A. The ACK response contains User B's SDP information.
6. On receiving a 200 OK response from User B, the call controller sends an ACK response to User B that contains User A's SDP information.

Third-party call control is often used for operator services (creating a call connecting two parties together) and conferencing.

The following call flows illustrate the use of third-party call control via Delayed Media support:

- SIP Gateway-to-SIP Gateway Call—Call Setup with Delayed Media via Third-Party Call Controller, page A-2
- SIP IP Phone-to-SIP Gateway—Call Setup and Call Hold with Delayed Media, page A-5
- SIP Gateway-to-uOne Call—Call Setup with Voice Mail, page A-9

Endpoints Specified as FQDNs in the SDP

With Cisco IOS Release 121(5)XM, SIP gateways can route on an FQDN in addition to routing on an IP address. When the SIP gateway receives a SIP message containing a FQDN in the c= SDP field, it parses the c= field of the SDP body, determines that it is a FQDN and resolves the next hop via a DNS query.

FigureA-4 illustrates a call flow of this feature.

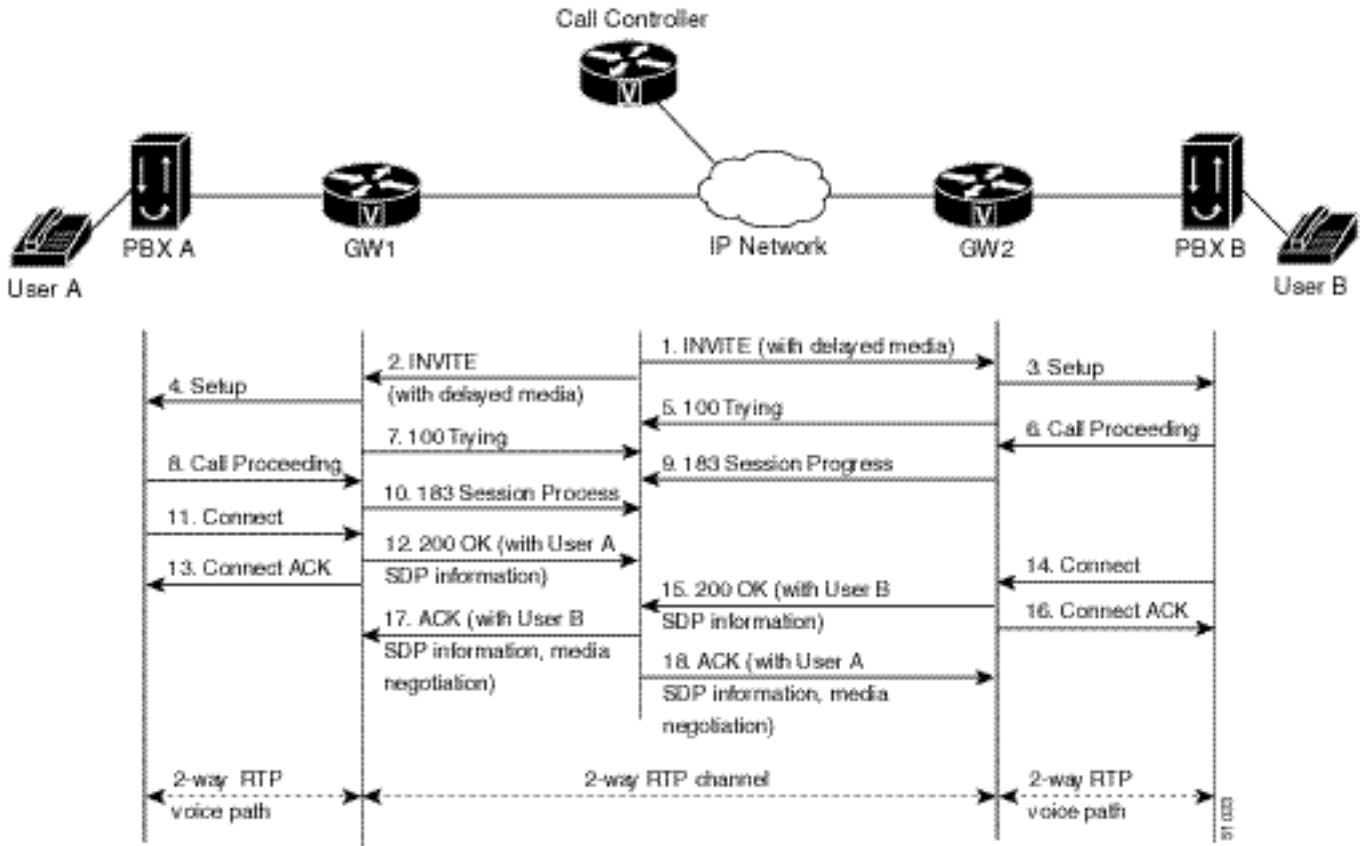
SIP Gateway-to-SIP Gateway Call—Call Setup with Delayed Media via Third-Party Call Controller

FigureA-1 illustrates a successful gateway-to-gateway call setup via a third-party call control agent.

The call flow scenario is as follows:

1. The call controller calls User B and then calls User A.
2. User A answers the call.
3. User B answers the call.
4. The call controller connects User A and User B.

FigureA-1 SIP Gateway-to-SIP Gateway—Call Setup via Third-Party Call Controller



Step	Action	Description
1	INVITE—Call controller to SIP gateway 2	<p>The third party call control agent sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter indicates that the Request-URI address is a telephone number rather than a user name. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The SDP media line is omitted or the INVITE does not contain any SDP information.

Step	Action	Description
2	INVITE—Call controller to SIP gateway 1	<p>The third party call control agent sends an INVITE request to SIP gateway 1. The INVITE request is an invitation to User A to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User A is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User A and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to User A appears as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter indicates that the Request-URI address is a telephone number rather than a user name. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The SDP media line is omitted or the INVITE does not contain any SDP information.
3	Setup—SIP gateway 2 to PBXB	SIP gateway 2 receives the INVITE request from the call controller and initiates a Call Setup with User B via PBX B.
4	Setup—SIP gateway 1 to PBXA	SIP gateway 1 receives the INVITE request from the call controller and initiates a Call Setup with User A via PBXA.
5	100 Trying—SIP gateway 2 to call controller	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by the call controller. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
6	Call Proceeding—PBX B to SIP gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
7	100 Trying—SIP gateway 1 to call controller	SIP gateway 1 sends a 100 Trying response to the INVITE request sent by the call controller. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User A has not yet been located and that some unspecified action, such as a database consultation, is taking place.
8	Call Proceeding—PBX A to SIP gateway 1	PBX A sends a Call Proceeding message to SIP gateway 1 to acknowledge the Call Setup request.
9	183 Session Progress—SIP gateway 2 to call controller	SIP gateway 2 sends a 183 Session Progress message to the call controller.
10	183 Session Progress—SIP gateway 1 to call controller	SIP gateway 1 sends a 183 Session Progress message to the call controller.
11	Connect—PBX A to SIP gateway 1	User A answers phone. PBX A sends a Connect message to SIP gateway 1. The Connect message notifies SIP gateway 1 that the connection has been made.
12	200 OK—SIP gateway 1 to call controller	<p>SIP gateway 1 sends a 200 OK response to the call controller. The 200 OK response notifies the call controller that the connection has been made.</p> <p>In the call 200 OK response, the SDP information of User A is specified.</p>

Step	Action	Description
13	Connect ACK—SIP gateway 1 to PBX A	SIP gateway 1 acknowledges PBX A's Connect message.
14	Connect—PBX B to SIP gateway 2	PBX B sends a Connect message to SIP gateway 2. The Connect message notifies SIP gateway 2 that the connection has been made.
15	200 OK—SIP gateway 2 to call controller	SIP gateway 2 sends a 200 OK response to the call controller. The 200 OK response notifies the call controller that the connection has been made. In the call 200 OK response, the SDP information of User B is specified.
16	Connect ACK—SIP gateway 2 to PBX B	SIP gateway 2 acknowledges PBX B's Connect message.
17	ACK—Call controller to SIP gateway 1	The call controller sends an ACK response to SIP gateway 1. The ACK response confirms that the call controller has received the 200 OK response from SIP gateway 1. In the ACK response, the SDP information of User B is specified. Media negotiation takes place.
14	ACK—Call controller to SIP gateway 2	The call controller sends an ACK response to SIP gateway 2. The ACK response confirms that the call controller has received the 200 OK response from SIP gateway 2. In the ACK response, the SDP information of User A is specified. Media negotiation takes place.

At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.

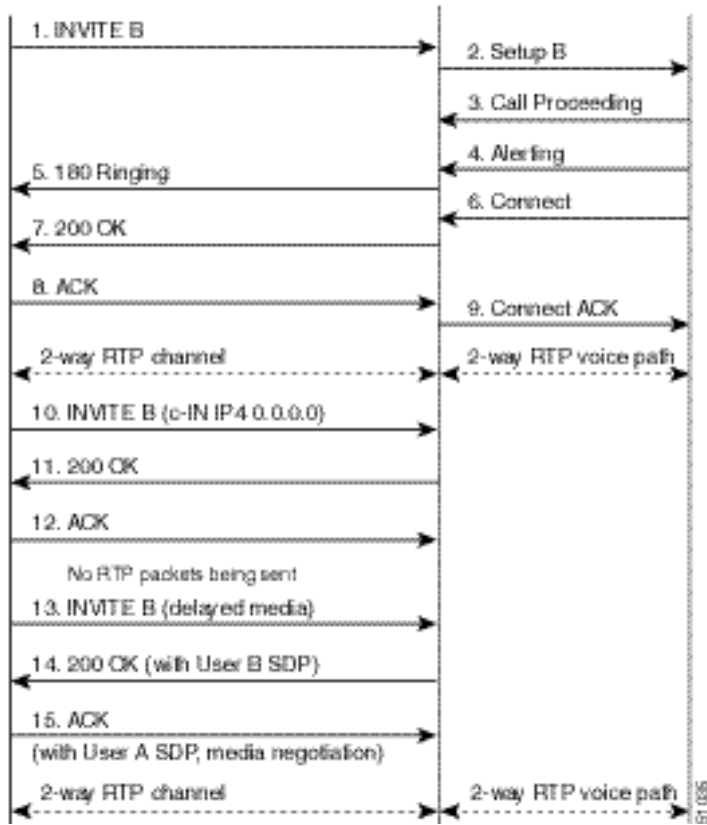
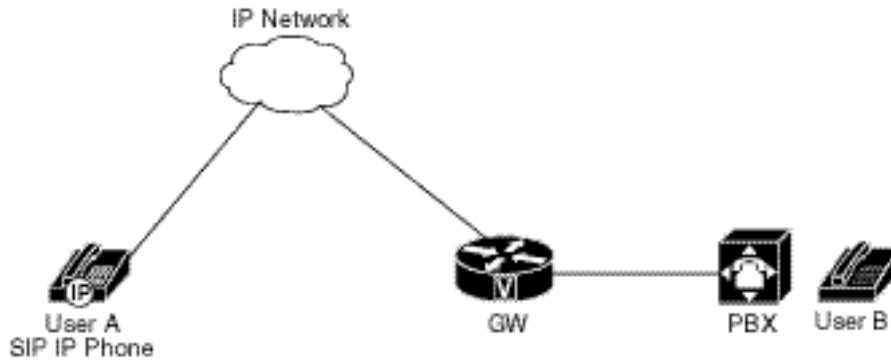
SIP IP Phone-to-SIP Gateway—Call Setup and Call Hold with Delayed Media

FigureA-2 illustrates a successful SIP IP phone-to-SIP gateway call setup and call hold using delayed media.

The call flow scenario is as follows:

1. User A calls User B.
2. User A puts User B on hold.
3. User A takes User B off hold.

Figure A-2 SIP IP Phone-to-SIP Gateway—Call Setup and Call Hold with Delayed Media



Step	Action	Description
1	INVITE—SIP IP phone to SIP gateway	<p>SIP IP phone sends an INVITE request to the SIP gateway. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter indicates that the Request-URI address is a telephone number rather than a user name. • SIP IP phone is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability User A is ready to receive is specified.
2	Setup B—SIP gateway to PBX	The SIP gateway receives the INVITE request from the call controller and initiates a Call Setup with User B via the PBX.
3	Call Proceeding—PBX to SIP gateway	The PBX sends a Call Proceeding message to the SIP gateway to acknowledge the Call Setup request.
4	Alerting—PBX to SIP gateway	The PBX sends an Alert message to the SIP gateway.
5	180 Ringing—SIP gateway to SIP IP phone	The SIP gateway sends a 180 Ringing response to the SIP IP phone. The 180 Ringing response indicates that the User B is being alerted.
6	Connect—PBX to SIP gateway	User B answers the phone. The PBX sends a Connect message to the SIP gateway. The Connect message notifies the SIP gateway that the connection has been made.
7	200 OK—SIP gateway to SIP IP phone	The SIP gateway sends a 200 OK response to the SIP IP phone. The 200 OK response notifies the SIP IP phone that the connection has been made.
8	ACK—SIP IP phone to SIP gateway	SIP IP phone sends an ACK to the SIP gateway. The ACK confirms that the SIP IP phone has received the 200 OK response from the SIP gateway.
9	Connect ACK—SIP gateway to PBX	The SIP gateway acknowledges the PBX’s Connect message. The call between User A and User B session is now active.
<p>A two-way RTP channel is established between the SIP IP phone and the SIP gateway. A two-way voice path is established between the SIP gateway and the PBX.</p>		
10	INVITE—SIP IP phone to SIP gateway	<p>SIP IP phone (User A) sends a mid-call INVITE request to the SIP gateway with a modified SDP session connection parameter (c=) that places the existing call between User A and User B on hold.</p> <p>The c= SDP field of the SIP INVITE contains an 0.0.0.0. This places the call in limbo.</p>

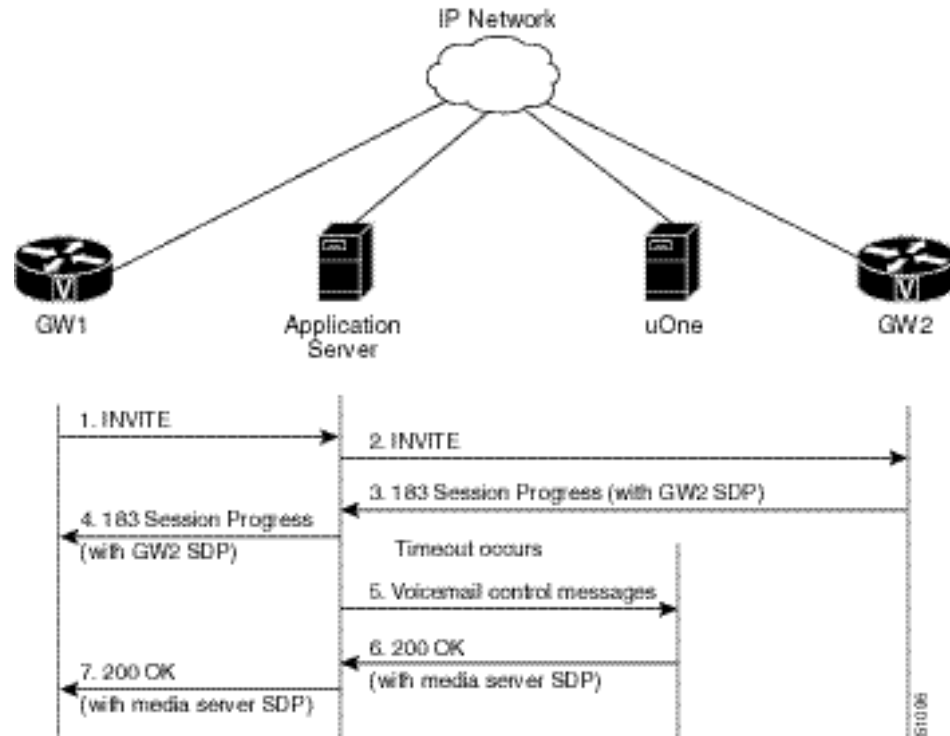
Step	Action	Description
11	200 OK—SIP gateway to SIP IP phone	SIP gateway 1 sends a 200 OK response to the SIP IP phone. The 200 OK response notifies the SIP IP phone that the INVITE was successfully processed.
12	ACK—SIP IP phone to SIP gateway	The SIP IP phone sends an ACK to the SIP gateway. The ACK confirms that the SIP IP phone has received the 200 OK response. The call session is now temporarily inactive. No RTP packets are being sent.
13	INVITE—SIP IP phone to SIP gateway	SIP IP phone sends an INVITE with delayed media to the SIP gateway. No media is specified in the SDP media name and transport address (m=) parameter.
14	200 OK—SIP gateway to SIP IP phone	The SIP gateway sends a 200 OK response to User A. In the 200 OK response, the SDP information of User B is specified.
15	ACK—SIP IP phone to SIP gateway	SIP IP phone sends an ACK to the SIP gateway. The ACK confirms that the SIP IP phone has received the 200 OK response from the SIP gateway. In the ACK response, the SDP information of User A is specified. Media negotiation takes place.

A two-way RTP channel is reestablished between the SIP IP phone and the SIP gateway. A two-way voice path (to User B) is established between the SIP gateway and the PBX.

SIP Gateway-to-uOne Call—Call Setup with Voice Mail

FigureA-3 illustrates a successful SIP gateway-to-uOne system call setup.

FigureA-3 SIP Gateway-to-SIP Gateway—Call Setup and Voice Mail



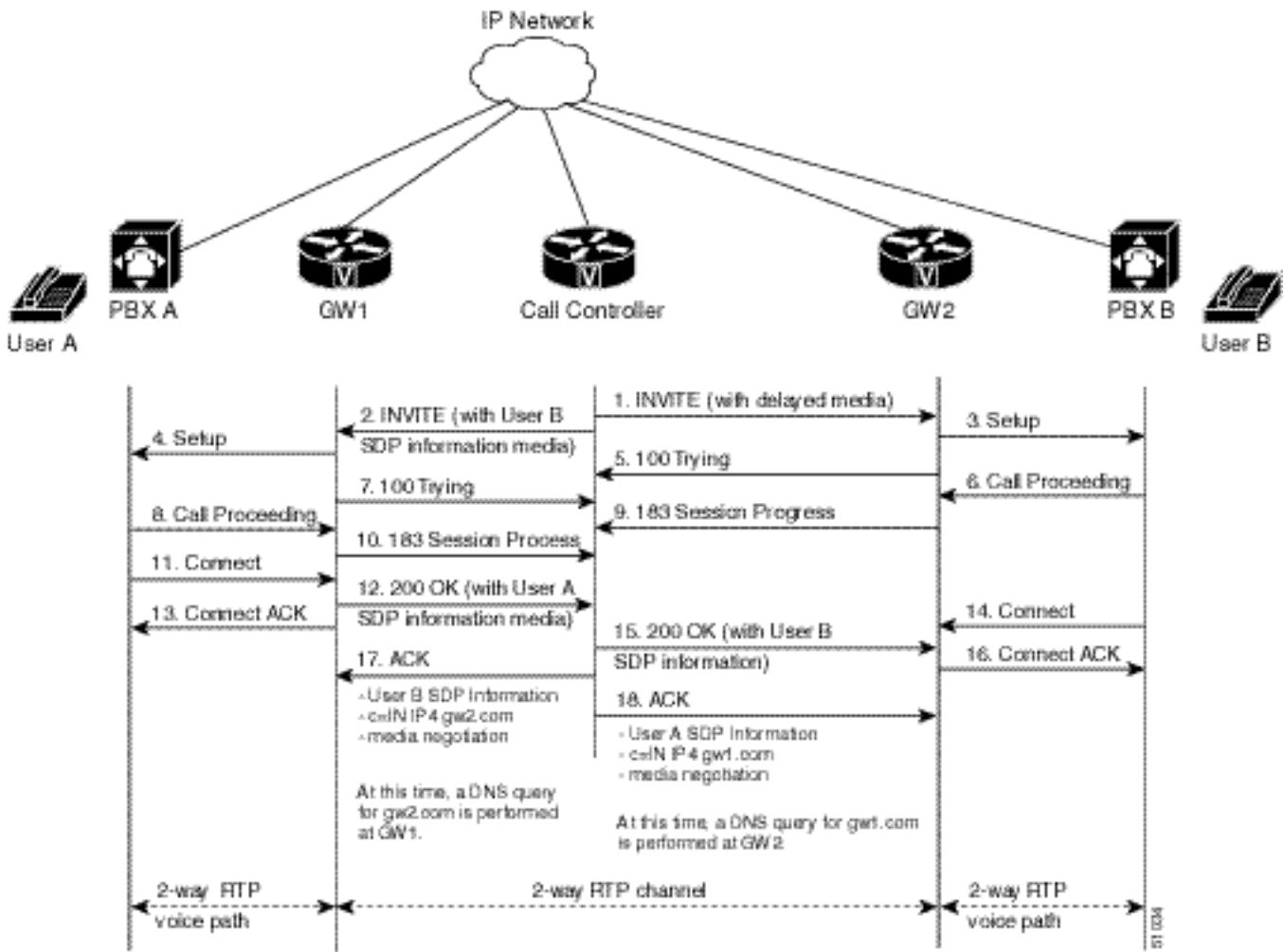
Step	Action	Description
1	INVITE—SIP gateway 1 to application server	SIP gateway 1 sends an INVITE request to the application server.
2	INVITE—Application server to SIP gateway 2	The application server sends the INVITE request to SIP gateway 2.
3	183 Session Progress—SIP gateway 2 to application server	SIP gateway 2 sends a 183 Session Progress message to the application server.
4	183 Session Progress—Application server to SIP gateway 1	The application server proxies the 183 Session Progress message to SIP gateway 1.
At this time a timeout occurs.		
5	Voice Mail control messages—Application server to uOne server	The application server forwards voice mail control messages to the uOne server (media server).
6	200 OK—uOne server to application server	The uOne server sends a 200 OK response to the application server. In the 200 OK response, the uOne server SDP is included.

Step	Action	Description
7	200 OK response—Application server to SIP gateway 1	The application server proxies the 200 OK response to the SIP gateway. In the 200 OK response, the uOne server SDP is included.

SIP Gateway-to-SIP Gateway—Call Setup using a FQDN and Delayed Media

FigureA-4 illustrates a successful gateway-to-gateway call setup using delayed media and a FQDN in the SDP session connection parameter.

FigureA-4 SIP Gateway-to-SIP Gateway—Call Setup using an FQDN in SDP



Step	Action	Description
1	INVITE—Call controller to SIP gateway 2	<p>The third party call control agent sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter indicates that the Request-URI address is a telephone number rather than a user name. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The SDP media line is omitted or the INVITE does not contain any SDP information.
2	INVITE—Call controller to SIP gateway 1	<p>The third party call control agent sends an INVITE request to SIP gateway 1. The INVITE request is an invitation to User A to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User A is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User A and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to User A appears as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter indicates that the Request-URI address is a telephone number rather than a user name. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The SDP media line is omitted or the INVITE does not contain any SDP information.
3	Setup—SIP gateway 2 to PBX B	SIP gateway 2 receives the INVITE request from the call controller and initiates a Call Setup with User B via PBX B.
4	Setup—SIP gateway 1 to PBX A	SIP gateway 1 receives the INVITE request from the call controller and initiates a Call Setup with User A via PBX A.
5	100 Trying—SIP gateway 2 to call controller	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by the call controller. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.

Step	Action	Description
6	Call Proceeding—PBX B to SIP gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
7	100 Trying—SIP gateway 1 to call controller	SIP gateway 1 sends a 100 Trying response to the INVITE request sent by the call controller. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User A has not yet been located and that some unspecified action, such as a database consultation, is taking place.
8	Call Proceeding—PBX A to SIP gateway 1	PBX A sends a Call Proceeding message to SIP gateway 1 to acknowledge the Call Setup request.
9	183 Session Progress—SIP gateway 2 to call controller	SIP gateway 2 sends a 183 Session Progress message to the call controller.
10	183 Session Progress—SIP gateway 1 to call controller	SIP gateway 1 sends a 183 Session Progress message to the call controller.
11	Connect—PBX A to SIP gateway 1	User A answers phone. PBX A sends a Connect message to SIP gateway 1. The Connect message notifies SIP gateway 1 that the connection has been made.
12	200 OK—SIP gateway 1 to call controller	SIP gateway 1 sends a 200 OK response to the call controller. The 200 OK response notifies the call controller that the connection has been made. In the call 200 OK response, the SDP information of User A is specified.
13	Connect ACK—SIP gateway 1 to PBX A	SIP gateway 1 acknowledges PBX A's Connect message.
14	Connect—PBX B to SIP gateway 2	PBX B sends a Connect message to SIP gateway 2. The Connect message notifies SIP gateway 2 that the connection has been made.
15	200 OK—SIP gateway 2 to call controller	SIP gateway 2 sends a 200 OK response to the call controller. The 200 OK response notifies the call controller that the connection has been made. In the call 200 OK response, the SDP information of User B is specified.
16	Connect ACK—SIP gateway 2 to PBX B	SIP gateway 2 acknowledges PBX B's Connect message.
17	ACK—Call controller to SIP gateway 1	The call controller sends an ACK response to SIP gateway 1. The ACK response confirms that the call controller has received the 200 OK response from SIP gateway 1. In the ACK response the media capability of User B is specified and the domain name of gateway 2 is specified in the SDP sessions parameter (c=) field. Media negotiation takes place. At this time, a DNS query is performed by SIP gateway 1 to resolve c=IN IP4 gw2.com.
18	ACK—Call controller to SIP gateway 2	The call controller sends an ACK to SIP gateway 2. The ACK confirms that the call controller has received the ACK response from SIP gateway 2. In the 200OK response the media capability of User A is specified and the domain name of gateway 1 is specified in the SDP sessions parameter (c=). Media negotiation takes place. At this time, a DNS query is performed by SIP gateway 2 to resolve c=INIP4gw1.com.

At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.



SIP Diversion Header Implementation for Redirecting Number

This appendix describes the SIP Diversion Header Implementation for Redirecting Number feature that is introduced in Cisco IOS Release 12.1(5)XM. In addition to the SIP Diversion Header Implementation for Redirecting Number feature, this appendix describes the SIP gateway ability to process a SIP 3xx Redirection response after the receipt of a SIP 18x Information response.

SIP Diversion Header Implementation for Redirecting Number

The SIP Diversion Header Implementation for Redirecting Number feature provides support for a new SIP header field; Call Control (CC)-Diversion. The CC-Diversion header field enables the SIP gateway to pass call control redirecting information during the call setup. Call control redirection is the redirection of a call based on a subscriber service such as call forwarding. Call redirection information is typically used for Unified Messaging and voice mail services to identify the recipient of a message. Call control redirection information can also be used to support applications such as automatic call distribution and enhanced telephony features such as Do Not Disturb and Caller ID.

If generated by the SIP gateway during call process, the CC-Diversion header field is based on the contents of the Redirecting Number Information Element (IE) in the ISDN Setup message. In addition, information such as the reason the call was redirected is included in the CC-Diversion header field.

FigureB-1 illustrates a call flow for this feature.

SIP Gateway 3xx Redirection Response Processing after 18x Information Responses

With Cisco IOS Release 12.1(5)XM, SIP gateways can process SIP 3xx Redirection responses after a 18x Information response has been received.

FigureB-2 illustrates the SIP gateway method of:

- Processing Redirecting Number IE in ISDN Setup messages
- Handling multiple CC-Diversion header fields
- Generating multiple CC-Diversion header fields

Gateway-to-Gateway Call—Redirection with CC-Diversion

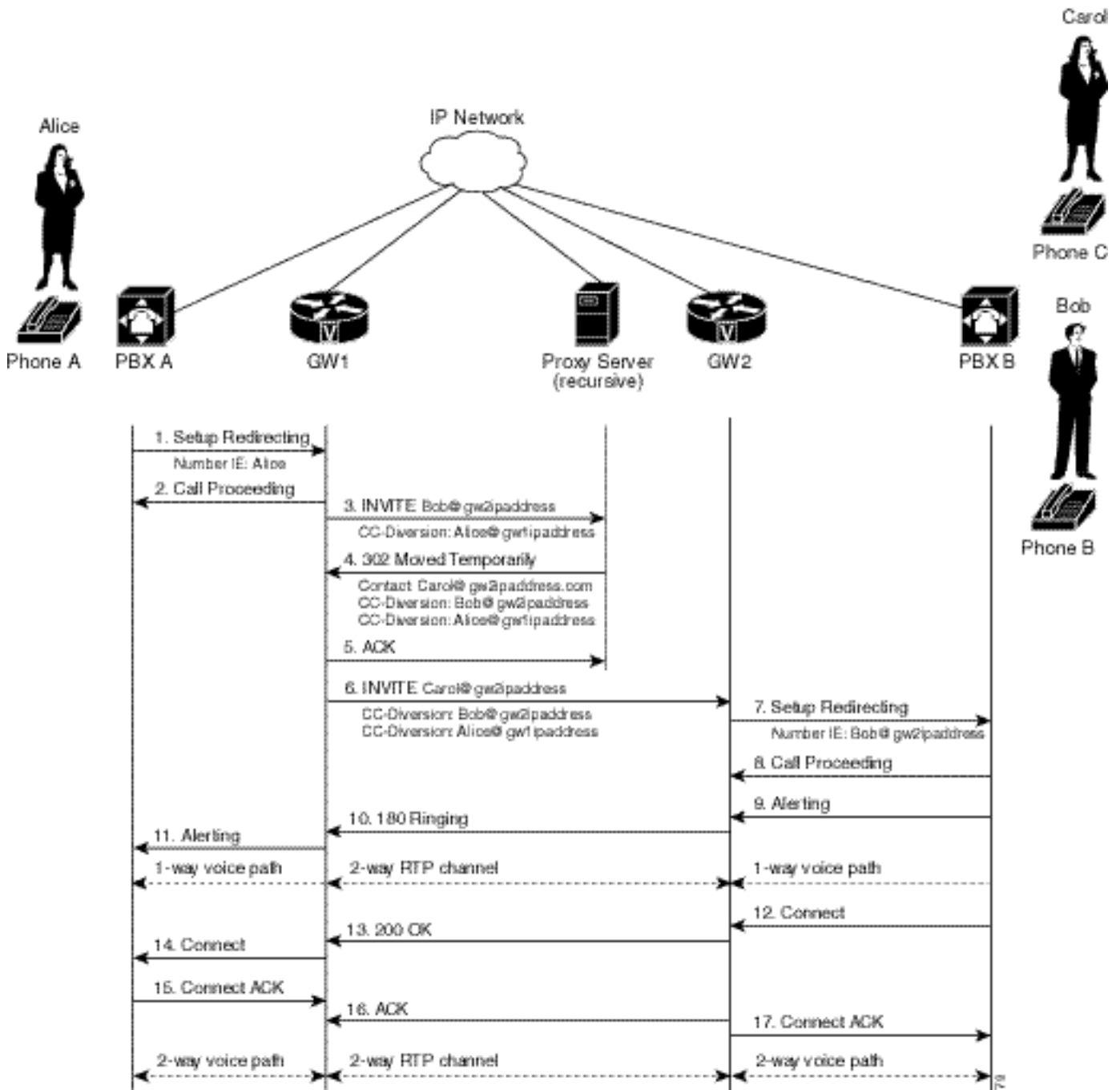
Figure B-1 illustrates a successful gateway-to-gateway call setup with call redirection with CC-Diversion.

In this scenario, the three end users are identified as Alice at phone A, Bob at phone B, and Carol at phone C. Alice at phone A is located at PBX A. PBX A is connected to SIP gateway 1 via a T1/E1. SIP gateway 1 is using a SIP redirect server. Bob at phone B and Carol at phone C are located at PBX B. PBX B is connected to SIP gateway 2 via a T1/E1. SIP gateway 1 is connected to SIP gateway 2 over an IP network.

The call flow scenario is as follows:

1. Bob at phone B has delegated his calls to Carol at phone C.
1. Alice at phone A initiates a call with Bob via SIP gateway 1 that is using a SIP proxy server.
2. The proxy server returns Carol at SIP gateway 2 as a contact location for Bob.
3. Gateway 1 initiates call with Carol.
4. Carol answers the call.

Figure B-1 SIP Gateway-to-SIP Gateway—Call Redirection with CC-Diversion



Step	Action	Description
1	Setup—PBX A to SIP gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as Alice at phone A attempts to call Bob at phone B.
2	Call Proceeding—SIP gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.

Step	Action	Description
3	INVITE—SIP gateway 1 to SIP proxy server	<p>GW1 sends an INVITE request to the proxy server. The INVITE request is an invitation to Bob to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • Bob’s phone number is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies Bob’s address and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to Bob might appear as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a user name. • Alice via GW1 is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability GW1 is ready to is specified in the SDP. • The port on which GW1 is prepared to receive the RTP data is specified in the SDP. • Alice at GW1 is specified as a CC-Diversion header. In addition, the CC-Diversion header field contains a reason for the diversion and a counter. Possible values for the diversion reason include the following: unknown, user-busy, no-answer, unconditional, deflection, and follow-me.
4	302 Moved Temporarily—SIP proxy server to SIP gateway 1	<p>The SIP proxy server determines that Bob’s calls have been configured to be redirected to Carol at phone C via GW2. The SIP proxy server sends an 302 Moved Temporarily message to GW1.</p> <p>In the 302 Moved Temporarily message, Carol at GW2 is added as the Contact and there are two CC-Diversion header fields included; one for Bob at GW2 (IP address or domain name) and one for Alice at GW1 (IP address or domain name).</p>
5	INVITE—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends an INVITE request to Carol at phone C via GW2. The INVITE request contains two CC-Diversion headers; one for Bob at GW2 (IP address or domain name) and one for Alice at GW1 (IP address or domain name).
6	Setup—SIP gateway 2 to PBXB	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates a Call Setup with Carol via PBX B. In the Call Setup, the ISDN Redirecting Number IE is generated using the contents of the top CC-Diversion header field; in this case, Bob at GW2.
7	Call Proceeding—PBX B to SIP gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.
8	Alerting—PBX B to SIP gateway 2	PBX B locates Carol at phone C and sends an Alert message to SIP gateway 2. Carol’s phone begins to ring.

Step	Action	Description
9	180 Ringing—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that SIP gateway 2 has located, and is trying to alert, Carol at phone C.
10	Alerting—SIP gateway 1 to PBX A	SIP gateway 1 sends an Alert message to PBX A. Alice hears ringback tone.
At this point, a one-way voice path is established between SIP gateway 1 and PBXA and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.		
11	Connect—PBX B to SIP gateway 2	Carol answers phone. PBX B sends a Connect message to SIP gateway 2. The Connect message notifies SIP gateway 2 that the connection has been made.
12	200 OK—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made. If Carol via SIP gateway 2 supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and Alice’s media capability in the 200 OK response. If Carol at SIP gateway 2 does not support the media capability advertised by Alice at SIP gateway 1, SIP gateway 2 sends back a 400 Bad Request response with a “Warning: 304 Codec negotiation failed” header field.
13	Connect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
14	Connect ACK—PBX A to SIP gateway 1	PBX A acknowledges SIP gateway 1’s Connect message.
15	ACK—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that the 200OK response has been received. The call is now in progress over a two-way voice path via RTP.
16	Connect ACK—SIP gateway 2 to PBX B	SIP gateway 2 acknowledges PBX B’s Connect message.

At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBXB. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.

Gateway-to-Gateway Call—SIP 3xx Redirection Response After Receipt of a SIP 18x Information Response

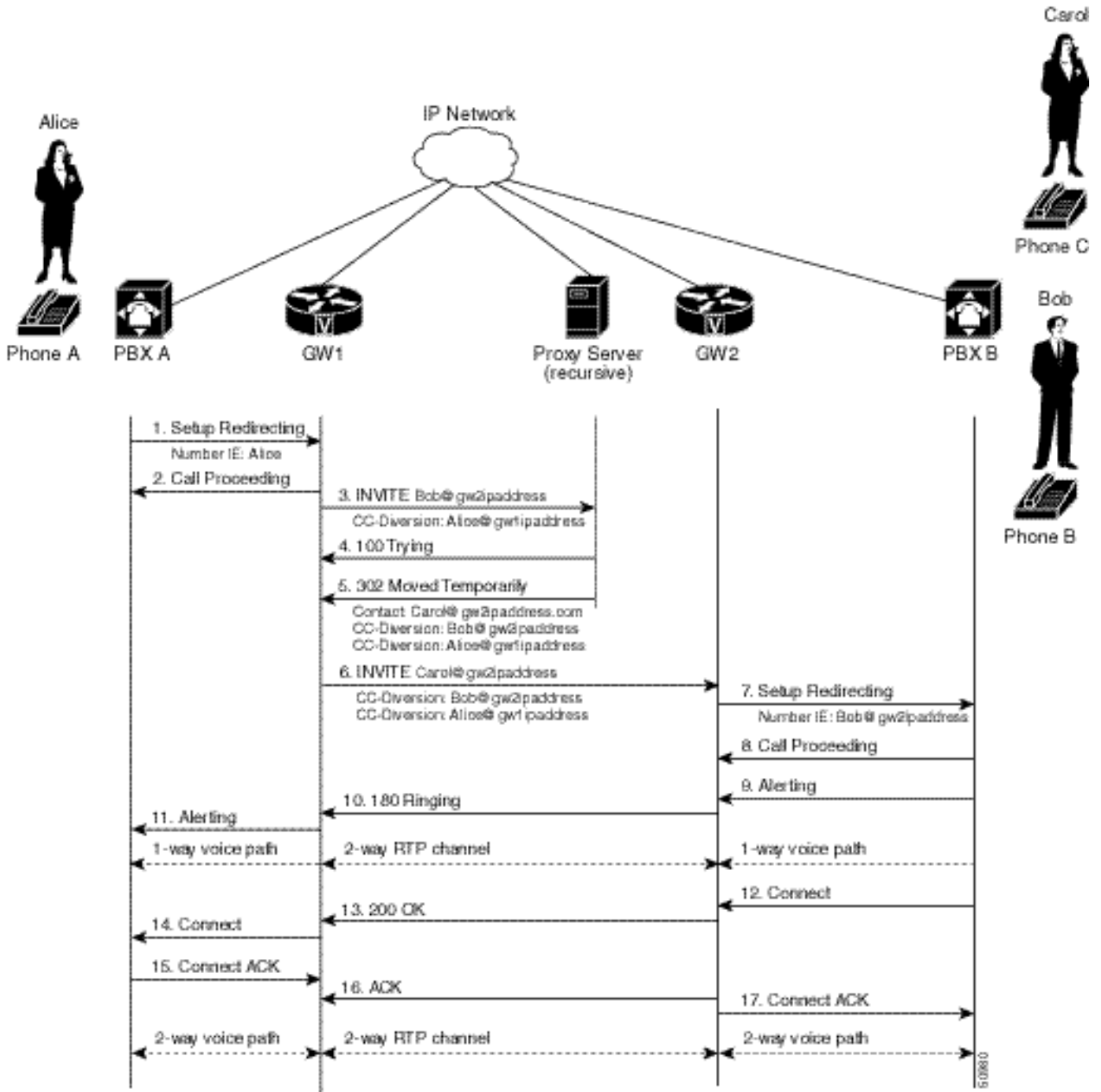
FigureB-2 illustrates a successful gateway-to-gateway call setup in which a SIP 3xx Redirection response is processed after the receipt of a SIP 18x Information response.

In this scenario, the three end users are identified as Alice at phone A, Bob at phone B, and Carol at phone C. Alice at phone A is located at PBX A. PBX A is connected to SIP gateway 1 via a T1/E1. SIP gateway 1 is using a SIP redirect server. Bob at phone B and Carol at phone C are located at PBX B. PBX B is connected to SIP gateway 2 via a T1/E1. SIP gateway 1 is connected to SIP gateway 2 over an IP network.

The call flow scenario is as follows:

1. Bob at phone B has delegated his calls to Carol at phone C.
1. Alice at phone A initiates a call with Bob via SIP gateway 1 that is using a SIP proxy server.
2. The proxy server returns Carol at SIP gateway 2 as a contact location for Bob.
3. Gateway 1 initiates call with Carol.
4. Carol answers the call.

Figure B-2 Gateway-to-Gateway Call—Call Redirection after Receipt of a SIP 18x Response



Step	Action	Description
1	Setup—PBX A to SIP gateway 1	Call Setup is initiated between PBX A and SIP gateway 1. The Call Setup includes the standard transactions that take place as Alice at phone A attempts to call Bob at phone B.

Step	Action	Description
2	Call Proceeding—SIP gateway 1 to PBX A	SIP gateway 1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
3	INVITE—SIP gateway 1 to SIP proxy server	<p>GW1 sends an INVITE request to the proxy server. The INVITE request is an invitation to Bob to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • Bob’s phone number is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies Bob’s address and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). For example, the Request-URI field in the INVITE request to Bob might appear as “INVITE sip:555-0002@companyb.com; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a user name. • Alice via GW1 is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability GW1 is ready to is specified in the SDP. • The port on which GW1 is prepared to receive the RTP data is specified in the SDP. • Alice at GW1 is specified as a CC-Diversion header. In addition, the CC-Diversion header field contains a reason for the diversion and a counter. Possible values for the diversion reason include the following: unknown, user-busy, no-answer, unconditional, deflection, and follow-me.
4	183 Session Progress—SIP proxy server to SIP gateway 1	The SIP proxy server sends a 183 Session Progress response to SIP gateway 1.
5	302 Moved Temporarily—SIP proxy server to SIP gateway 1	<p>The SIP proxy server determines that Bob’s calls have been configured to be redirected to Carol at phone C via GW2. The SIP proxy server sends an 302 Moved Temporarily message to GW1.</p> <p>In the 302 Moved Temporarily message, Carol at GW2 is added as the Contact and there are two CC-Diversion header fields included; one for Alice at GW1 (IP address or domain name) and Bob at GW2 (IP address or domain name).</p>
6	INVITE—SIP gateway 1 to SIP gateway 2	SIP gateway 1 sends an INVITE request to Carol at phone C via GW2. The INVITE request contains two CC-Diversion headers; one for Alice at GW1 (IP address or domain name) and Bob at GW2 (IP address or domain name).
7	Setup—SIP gateway 2 to PBXB	SIP gateway 2 receives the INVITE request from SIP gateway 1 and initiates a Call Setup with Carol via PBX B. In the Call Setup, the ISDN Redirecting Number IE is generated using the contents of the top CC-Diversion header field; in this case, Bob at GW2.
8	Call Proceeding—PBX B to SIP gateway 2	PBX B sends a Call Proceeding message to SIP gateway 2 to acknowledge the Call Setup request.

Step	Action	Description
9	Alerting—PBX B to SIP gateway 2	PBX B locates Carol at phone C and sends an Alert message to SIP gateway 2. Carol's phone begins to ring.
10	180 Ringing—SIP gateway 2 to SIP gateway 1	SIP gateway 2 sends a 180 Ringing response to SIP gateway 1. The 180 Ringing response indicates that SIP gateway 2 has located, and is trying to alert, Carol at phone C.
11	Alerting—SIP gateway 1 to PBX A	SIP gateway 1 sends an Alert message to PBX A. Alice hears ringback tone.
<p>At this point, a one-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>		
12	Connect—PBX B to SIP gateway 2	Carol answers phone. PBX B sends a Connect message to SIP gateway 2. The Connect message notifies SIP gateway 2 that the connection has been made.
13	200 OK—SIP gateway 2 to SIP gateway 1	<p>SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made.</p> <p>If Carol via SIP gateway 2 supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and Alice's media capability in the 200 OK response. If Carol at SIP gateway 2 does not support the media capability advertised by Alice at SIP gateway 1, SIP gateway 2 sends back a 400 Bad Request response with a "Warning: 304 Codec negotiation failed" header field.</p>
14	Connect—SIP gateway 1 to PBX A	SIP gateway 1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
15	Connect ACK—PBX A to SIP gateway 1	PBX A acknowledges SIP gateway 1's Connect message.
16	ACK—SIP gateway 1 to SIP gateway 2	<p>SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that the 200OK response has been received.</p> <p>The call is now in progress over a two-way voice path via RTP.</p>
17	Connect ACK—SIP gateway 2 to PBX B	SIP gateway 2 acknowledges PBX B's Connect message.
<p>At this point, a two-way voice path is established between SIP gateway 1 and PBX A and between SIP gateway 2 and PBX B. A two-way RTP channel is established between SIP gateway 1 and SIP gateway 2.</p>		



Troubleshooting Tips for Call Flow Scenarios

This document shows the **debug ccsip messages** command traces the SIP messages exchanges between the SIP User Agent client and the gateway in the Cisco IOS Release 12.2(2)XB and Cisco IOS Release 12.2(8)T, including examples and explanations of the output generated.

- SIP Call with Diversion Header Added for Call-Diversion Output from GW1 Side, page C-7
- SIP Call with RSVP QoS Output from GW1 Side, page C-13
- SIP Call Using RFC2833 for DTMF-Relay Output from GW1 Side, page C-21
- SIP Provisional Response Output from GW1 Side, page C-27
- SIP T.38 Call with QoS Enabled Output, page C-32




Note

For an explanation of SIP functions, methods, and responses see “*Cisco SIP Compliance Reference Information*”.

SIP Header Fields

Header Field	Definition
Call-ID	The Call-ID general-header field uniquely identifies a specific invitation or all registrations of a specific client. Note that a single multimedia conference can give rise to several calls with different Call-IDs. For example, if a user invites a single individual several times to the same (long-running) conference.
Contact	The Contact general-header field can appear in INVITE, ACK, and REGISTER requests, and in 1xx, 2xx, 3xx, and 485 responses. In general, it provides a URL where the user can be reached for further communications.
Content-Length	The Content-Length entity-header field indicates the size of the message-body, in decimal number of octets, sent to the recipient.
Content-Type	The Content-Type entity-header field indicates the media type of the message-body sent to the recipient.

Header Field	Definition
Cseq	<p>Users MUST add the CSeq (command sequence) general-header field to every request. A CSeq header field in a request contains the request method and a single decimal sequence number chosen by the requesting client, unique within a single value of Call-ID. The sequence number MUST be expressed as a 32-bit unsigned integer. The initial value of the sequence number is arbitrary, but MUST be less than 2^{31}. Consecutive requests that differ in request method, headers, or body, but have the same Call-ID MUST contain strictly monotonically increasing and contiguous sequence numbers; sequence numbers do not wrap around. Retransmissions of the same request carry the same sequence number, but an INVITE with a different message body or different header fields (a “re-invitation”) acquires a new, higher sequence number. A server MUST echo the CSeq value from the request in its response. If the Method value is missing in the received CSeq header field, the server fills it in appropriately.</p>
Date	<p>Date is a general-header field. Its syntax is:</p> <p>SIP-date = rfc1123-date</p> <p>Note that unlike HTTP/1.1, SIP only supports the most recent RFC 1123 [29] formatting for dates.</p>
Diversion	<p> Note Currently the CC-Diversion header is used but it will change to just Diversion. Values will not change though.</p>

Header Field	Definition
Expires	<p>The Expires entity-header field gives the date and time after which the message content expires.</p> <p>This header field is currently defined only for the REGISTER and INVITE methods. For REGISTER, it is a request and response-header field. In a REGISTER request, the client indicates how long it wants the registration to be valid. In the response, the server indicates the earliest expiration time of all registrations. The server MAY choose a shorter time interval than that requested by the client, but SHOULD NOT choose a longer one.</p>
From	<p>Requests and responses MUST contain a From general-header field, indicating the initiator of the request. The From field MAY contain the “tag” parameter. The server copies the From header field from the request to the response. The optional “display-name” is meant to be rendered by a human-user interface. A system SHOULD use the display name “Anonymous” if the identity of the client is to remain hidden.</p> <p>The SIP-URL MUST NOT contain the “transport-param”, “maddr-param”, “ttl-param”, or “headers” elements. A server that receives a SIP-URL with these elements removes them before further processing.</p>

Header Field	Definition
Max-Forwards	<p>The Max-Forwards request-header field may be used with any SIP method to limit the number of proxies or gateways that can forward the request to the next downstream server. This can also be useful when the client is attempting to trace a request chain which appears to be failing or looping in mid chain.</p> <p>The Max-Forwards value is a decimal integer indicating the remaining number of times this request message is allowed to be forwarded.</p> <p>Each proxy or gateway recipient of a request containing a Max-Forwards header field MUST check and update its value before forwarding the request. If the received value is zero (0), the recipient MUST NOT forward the request. Instead, for the OPTIONS and REGISTER methods, it MUST respond as the final recipient. For all other methods, the server returns 483 (too many hops).</p> <p>If the received Max-Forwards value is greater than zero, then the forwarded message MUST contain an updated Max-Forwards field with a value decremented by one (1).</p>
Require	<p>The Require request-header field is used by clients to tell useragent servers about options that the client expects the server to support in order to properly process the request. If a server does not understand the option, it MUST respond by returning status code 420 (bad extension) and list those options it does not understand in the Unsupported header.</p>

Header Field	Definition
Server	The Server response-header field contains information about the software used by the user agent server to handle the request.
Timestamp	The timestamp general-header field describes when the client sent the request to the server. The value of the timestamp is of significance only to the client and it MAY use any time scale. The server MUST echo the exact same value and MAY, if it has accurate information about this, add a floating point number indicating the number of seconds that have elapsed since receiving the request. The timestamp is used by the client to compute the round-trip time to the server so that it can adjust the time out value for retransmissions.
To	<p>The To general-header field specifies recipient of the request, with the same SIP URL syntax as the From field.</p> <p>Requests and responses MUST contain a To general-header field, indicating the desired recipient of the request. The optional “display-name” is meant to be rendered by a human-user interface. The UAS or redirect server copies the To header field into its response, and MUST add a “tag” parameter if the request contained more than one Via header field.</p>
User-Agent	The User-Agent general-header field contains information about the client user agent originating the request.
Via	The Via field indicates the path taken by the request so far. This prevents request looping and ensures replies take the same path as the requests, which assists in firewall traversal and other unusual routing situations.

SIP Call with Diversion Header Added for Call-Diversion Output from GW1 Side

RS = Redirect Server

```
GW1 ---INVITE---> RS (F1)
GW1 <----302----- RS (F2)
GW1 -----ACK-----> RS (F3)
GW1 ---INVITE---> GW2 (F4)
GW1 <----100----- GW2 (F5)
GW1 <----183----- GW2 (F6)
GW1 <----200OK----- GW2 (F7)
GW1 -----ACK-----> GW2 (F8)

GW1 <----BYE----- GW2 (F9)
GW1 ---200OK-----> GW2 (F10)
```

In this call the originator gets redirected to the terminator by a proxy. The proxy attaches the 'Diversion:' headers to the 3xx response that it sends to the originator. The 3xx response also has the contact header with the address of the terminator. This header is defined in IETF draft 'draft-levy-sip-diversion-02.txt', although the syntax is slightly different because of an implementation in IOS software before the draft being recognized by the IETF.

```
(F1)
Sent:
INVITE sip:3111100@64.102.17.80:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:3111100@64.102.17.80;user=phone>
Date: Mon, 01 Mar 1993 00:40:08 GMT65
Call-ID: 232226D2-14F411CC-80279792-19DC655A@172.18.193.98
Cisco-Guid: 557120379-351539660-2149947282-433874266
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 730946408
Contact: <sip:36601@172.18.193.98:5060;user=phone>
Expires: 180
Content-Type: application/sdp
Content-Length: 160

v=0
o=CiscoSystemsSIP-GW-UserAgent 4075 3837 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 16526 RTP/AVP 18
a=rtpmap:18 G729/8000
```

```
(F2)
Received:
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:3111100@64.102.17.80;user=phone>
Date: Mon, 01 Mar 1993 00:40:08 GMT
Call-ID: 232226D2-14F411CC-80279792-19DC655A@172.18.193.98
Cisco-Guid: 557120379-351539660-2149947282-433874266
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
CC-Diversion: <sip:36602@172.18.193.120>;reason=unconditional
CC-Diversion: <sip:33456@172.18.193.121>;reason=user-busy
CC-Diversion: <sip:36342@172.18.193.122>;reason=unconditional
CC-Diversion: <sip:23222@172.18.193.123>;reason=no-answer
Contact: Anonymous <sip:36602@172.18.193.120>
Contact: Anonymous <sip:36601@172.18.193.98>
```

**Note**

The Redirect Server sends back a 302 message, with a new Contact: address, and a Diversion: header added. The GW will now copy those headers into the new INVITE,

```
(F3)
00:40:08:
Sent:
ACK sip:3111100@64.102.17.80:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:3111100@64.102.17.80;user=phone>
Date: Mon, 01 Mar 1993 00:40:08 GMTCall-ID:
232226D2-14F411CC-80279792-19DC655A@172.18.193.98
Max-Forwards: 6
Content-Length: 0
CSeq: 101 ACK
(F4)
00:40:08: Sent:
INVITE sip:36602@172.18.193.120:5060 SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:3111100@64.102.17.80;user=phone>
Date: Mon, 01 Mar 1993 00:40:08 GMT
Call-ID: 23704587-14F411CC-80289792-19DC655A@172.18.193.98
Cisco-Guid: 557120379-351539660-2149947282-433874266
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 730946408
Contact: <sip:36601@172.18.193.98:5060;user=phone>
CC-Diversion: <sip:36602@172.18.193.120>;reason=unconditional
CC-Diversion: <sip:33456@172.18.193.121>;reason=user-busy
CC-Diversion: <sip:36342@172.18.193.122>;reason=unconditional
CC-Diversion: <sip:23222@172.18.193.123>;reason=no-answer
Expires: 180
Content-Type: application/sdp
Content-Length: 160
v=0
o=CiscoSystemsSIP-GW-UserAgent 9443 2845 IN IP4 172.18.193.98
s=SIP Callc=IN IP4 172.18.193.98
t=0 0m=audio 18116 RTP/AVP 18
a=rtpmap:18 G729/8000
```

```
(F5)
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:3111100@64.102.17.80;user=phone>
Date: Mon, 01 Mar 1993 00:40:09 GMT
Call-ID: 23704587-14F411CC-80289792-19DC655A@172.18.193.98
Timestamp: 730946408
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITEContent-Length: 0
(F6)
```

```
Received:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:3111100@64.102.17.80;user=phone>
Date: Mon, 01 Mar 1993 00:40:09 GMT
Call-ID: 23704587-14F411CC-80289792-19DC655A@172.18.193.98
```

```

Timestamp: 730946408
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Content-Type: application/sdp
Session: Media
Content-Length: 162
v=0
o=CiscoSystemsSIP-GW-UserAgent 2626 2857 IN IP4 172.18.193.120
s=SIP Call
c=IN IP4 172.18.193.120
t=0 0
m=audio 19030 RTP/AVP 18
a=rtpmap:18 G729/8000

```

```

(F7)
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:3111100@64.102.17.80;user=phone>;tag=24DE3C-1EBF
Date: Mon, 01 Mar 1993 00:40:09 GMT
Call-ID: 23704587-14F411CC-80289792-19DC655A@172.18.193.98
Timestamp: 730946408
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:36602@172.18.193.120:5060;user=phone>
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 162
v=0
o=CiscoSystemsSIP-GW-UserAgent 2626 2857 IN IP4 172.18.193.120
s=SIP Call
c=IN IP4 172.18.193.120
t=0 0
m=audio 19030 RTP/AVP 18
a=rtpmap:18 G729/8000

```

```

(F8)
Sent:
ACK sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:3111100@64.102.17.80;user=phone>;tag=24DE3C-1EBF
Date: Mon, 01 Mar 1993 00:40:08 GMT
Call-ID: 23704587-14F411CC-80289792-19DC655A@172.18.193.98
Max-Forwards: 6
Content-Length: 0CSeq: 101 ACK

```

```

(F9)
Received:
BYE sip:36601@172.18.193.98:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.120:5060
From: <sip:3111100@64.102.17.80;user=phone>;tag=24DE3C-1EBF
To: "36601"<sip:36601@172.18.193.98>
Date: Mon, 01 Mar 1993 00:40:16 GMT
Call-ID: 23704587-14F411CC-80289792-19DC655A@172.18.193.98
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 730946416
CSeq: 101 BYE
Content-Length: 0

```

```

(F10)
00:40:15: Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.120:5060
From: <sip:3111100@64.102.17.80;user=phone>;tag=24DE3C-1EBF

```

```
To: "36601"<sip:36601@172.18.193.98>
Date: Mon, 01 Mar 1993 00:40:15 GMT
Call-ID: 23704587-14F411CC-80289792-19DC655A@172.18.193.98
Server: Cisco-SIPGateway/IOS-12.x
Timestamp: 730946416
Content-Length: 0
CSeq: 101 BYE
```

FigureC-1 SIP Gateway to SIP Gateway — Call Redirection with Diversion

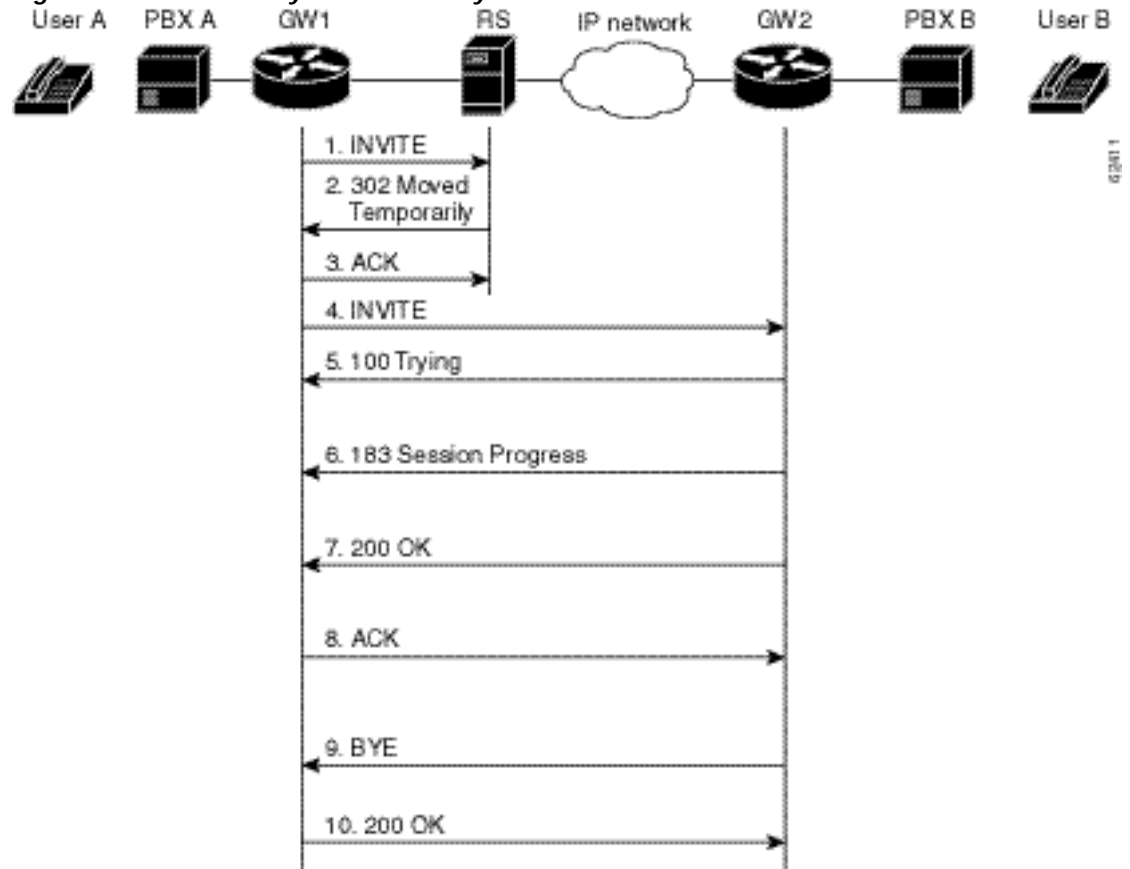


Table C-1 SIP Call with Diversion Header Added for Call-Diversion Output from GW1 Side

Step	Action	Description
F1	INVITE—SIP Gateway 1 to SIP Redirect Server	<p>SIP gateway 1 sends an INVITE request to the SIP redirect server. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). In this example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:3111100@64.102.17.80:5060; user=phone”. The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a username. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability (designated by m=) that User A is ready to receive is specified. In this case m=audio 16526 RTP/AVP 18. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified in the m= line.
F2	302 Moved Temporarily—SIP Redirect Server to SIP Gateway 1	<p>In this example the SIP redirect server sends a 302 Moved Temporarily response to SIP gateway 1. The 302 Moved Temporarily response indicates that the SIP redirect server accepted the INVITE request, contacted a location server with all or part of User B’s SIP URL, and the user was no longer available at the specified location. The server provided an alternate location for User B in the header. This response indicates that the user is no longer available at the specified location. An alternate location is included in the Contact: header. The SIP redirect server returns the alternate address to SIP gateway 1 in the 302 Moved Temporarily response.</p>
F3	ACK—SIP Gateway 1 to SIP Redirect Server	<p>SIP gateway 1 acknowledges the 302 Moved Temporarily response with an ACKnowledgement (ACK).</p>
F4	INVITE—SIP Gateway 1 to SIP Gateway 2	<p>SIP gateway 1 sends a new INVITE request to SIP gateway 2. The new INVITE request includes the first contact listed in the 302 Moved Temporarily response as the new address gher transaction number in the CSeq field, and the same Call-ID as the first INVITE request.</p>
F5	100 Trying—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located.</p>
F6	183 Session Progress—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 sends a 183 Session Progress response to SIP gateway 1. The 183Session Progress response indicates that SIP gateway 2 has located, and is trying to perform inband alerting for the caller.</p>

Step	Action	Description
F7	200 OK—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made.</p> <p>If User B supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.</p>
F8	ACK—SIP Gateway 1 to SIP Gateway 2	<p>SIP gateway 1 sends an ACKnowledge (ACK) to SIP gateway 2. The ACK confirms that the 200 OK response has been received.</p> <p>The call is now in progress over a two-way voice path via RTP.</p>
F9	BYE—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
F10	200 OK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.

SIP Call with RSVP QoS Output from GW1 Side

```

GW1 ---INVITE---> GW2 (F1)
GW1 <-100Trying-- GW2 (F2)
GW1 <--183/QoS--- GW2 (F3)
GW1 ---PRACK----> GW2 (F4)
GW1 <-200/PRACK-- GW2 (F5)
GW1 ---COMET----> GW2 (F6)
GW1 <-200/COMET-- GW2 (F7)
GW1 <-183SesProg- GW2 (F8)
GW1 ---PRACK----> GW2 (F9)
GW1 <-200/PRACK-- GW2 (F10)
GW1 <--200/INV--- GW2 (F11)
GW1 -----ACK----> GW2 (F12)
GW1 -----BYE----> GW2 (F13)
GW1 <----200OK---- GW2 (F14)

```

**Note**

The call-flows and messaging is per the IETF draft 'draft-ietf-sip-manyfolks-resource-02.txt'. They show the GW integrating the messaging for SIP to use RSVP to signal the network for resources prior to a VoIP call being setup.

```
(F1)
Sent:
INVITE sip:36602@172.18.193.120;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>
Date: Mon, 01 Mar 1993 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
Supported: 100rel
Cisco-Guid: 2083424384-351343052-2148441368-2058520395
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 730945271
Contact: <sip:36601@172.18.193.98:5060;user=phone>
Expires: 180
Content-Type: application/sdp
Content-Length: 211

v=0
o=CiscoSystemsSIP-GW-UserAgent 5800 3094 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 17158 RTP/AVP 0 18
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=qos:mandatory sendrecv
```

**Note**

INVITE sent with QoS required. This is signaled in the Supported: header.

```
(F2)
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 00:20:42 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
Timestamp: 730945271
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Content-Length: 0
```

```
(F3)
Received:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 00:20:42 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
Timestamp: 730945271
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:36602@172.18.193.120:5060;user=phone>
CSeq: 101 INVITE
Require: 100rel
RSeq: 8044
Content-Type: application/sdp
Content-Disposition: precondition;handling=required
Content-Length: 196
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 1431 3084 IN IP4 172.18.193.120
s=SIP Call
c=IN IP4 172.18.193.120
t=0 0
m=audio 17030 RTP/AVP 18
a=rtpmap:18 G729/8000
a=qos:mandatory sendrecv confirm
```

**Note**

Response received with the QoS request acknowledged.

```
(F4)
Sent:
PRACK sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
CSeq: 102 PRACK
RAck: 8044 101 INVITE
Content-Length: 0
```

**Note**

Acknowledgement of the QoS request is received.

```
(F5)Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 00:20:42 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 102 PRACK
Content-Length: 0
```

```
(F6)Sent:
COMET sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
CSeq: 103 COMET
Content-Type: application/sdp
Content-Length: 209
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 5800 3094 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 17158 RTP/AVP 0 18
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=qos:success sendrecv
```

**Note**

Bidirectional QoS is established. The RSVP signaling between the gateways and the IP network has been completed as well.

```
(F7)
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 00:20:42 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 103 COMET
Content-Length: 0
```

```
(F8)
Received:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 00:20:42 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
Timestamp: 730945271
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:36602@172.18.193.120:5060;user=phone>
CSeq: 101 INVITE
Require: 100rel
RSeq: 8045
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 162
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 1431 3084 IN IP4 172.18.193.120
s=SIP Call
c=IN IP4 172.18.193.120
t=0 0
m=audio 17030 RTP/AVP 18
a=rtpmap:18 G729/8000
```

```
(F9)
Sent:
PRACK sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
CSeq: 104 PRACK
RAck: 8045 101 INVITE
Content-Length: 0
```

```
(F10)
Received: SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
```

```
Date: Mon, 01 Mar 1993 00:20:42 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 104 PRACK
Content-Length: 0
```

```
(F11)
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 00:20:42 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
Timestamp: 730945271
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:36602@172.18.193.120:5060;user=phone>
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 162
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 1431 3084 IN IP4 172.18.193.120
s=SIP Call
c=IN IP4 172.18.193.120
t=0 0
m=audio 17030 RTP/AVP 18
a=rtpmap:18 G729/8000
```

```
(F12)
Sent:
ACK sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
Max-Forwards: 6
Content-Length: 0
CSeq: 101 ACK
```

```
(F13)
Sent:
BYE sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
Date: Mon, 01 Mar 1993 GMT
Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 730945310
CSeq: 105 BYE
Content-Length: 0
```

```
(F14)Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=136564-4EF
To: <sip:36602@172.18.193.120;user=phone>;tag=12F5AC-268A
```

Date: Mon, 01 Mar 1993 00:21:22 GMT
 Call-ID: 7D5FB580-14F111CC-80109D18-7AB2874B@172.18.193.98
 Server: Cisco-SIPGateway/IOS-12.x
 Timestamp: 730945310
 Content-Length: 0

FigureC-2 SIP Call with RSVP QoS

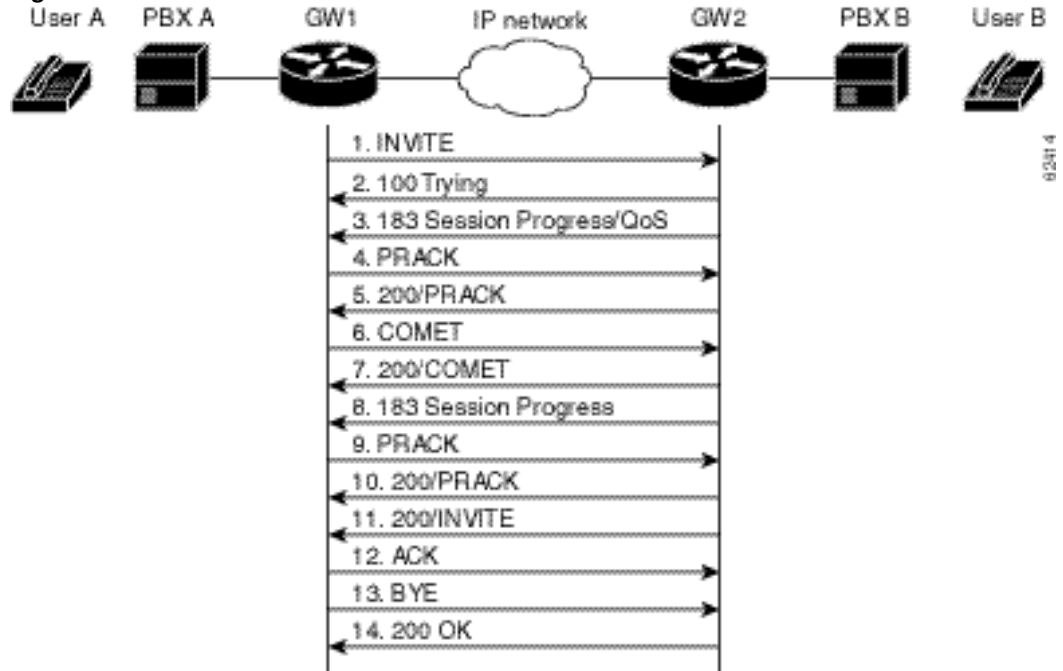


Table C-2 SIP Call with RSVP QoS Output from GW1 Side

Step	Action	Description
F1	INVITE—SIP Gateway 1 to SIP Gateway 2	<p>SIP gateway 1 sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <p>The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an email address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). In this example, the Request-URI field in the INVITE request to User B appears as “36602@172.18.193.120;user=phone”. The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a username.</p> <ul style="list-style-type: none"> • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability (designated by m=) User A is ready to receive is specified. In this case m=audio 17158 RTP/AVP 0 18. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
F2	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located.
F3	183 Session Progress/QoS—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 sends a 183 Session Progress response to SIP gateway 1. The 183 Session Progress response indicates that SIP gateway 2 has located, and is trying to perform inband alerting for the caller.</p> <p>Additionally Quality of Service (QoS) reserves sufficient network-layer resources to guarantee bandwidth and bounds on packet loss, delay, and jitter; thus ensuring that the called party's phone rings only after bandwidth required for the call has been successfully reserved.</p>
F4	PRACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 acknowledges the 183 Session Progress/QoS response with a PRovisional ACKnowledgement (PRACK).
F5	200/PRACK—SIP Gateway 2 to SIP Gateway 1	<p>The 200OK/PRovisional ACKnowledgement (PRACK) response notifies SIP gateway 1 that the connection has been made.</p> <p>The SIP gateway generates this response when the PBX indicates that the connection has been made. On receiving this response, the gateway forwards the response to the corresponding party and responds with a PRACK.</p>
F6	COMET—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a COnditions MET (COMET) response to SIP gateway 2 indicating that the preconditions have been met.
F7	200/COMET—SIP Gateway 2 to SIP Gateway 1	The 200 OK/COnditions MET (COMET) response notifies SIP gateway 1 that the conditions for the call have been met.

Step	Action	Description
F8	183 Session Progress— SIP Gateway 2 to SIP Gateway 1	The 183 Session Progress response is used to perform inband alerting for the caller.
F9	PRACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 acknowledges the 183 Session Progress response with a PRovisional ACKnowledgement (PRACK).
F10	200/PRACK—SIP Gateway 2 to Gateway 1	The 200OK/PRovisional ACKnowledgement (PRACK) response notifies SIP gateway 1 that the connection has been made. The SIP gateway generates this response when the PBX indicates that the connection has been made. Upon receiving this response, the gateway forwards the response to the corresponding party and responds with a PRACK.
F11	200/INVITE—SIP Gateway 2 to SIP Gateway 1	The 200/INVITE response acknowledges that the connection has been made and SIP gateway 2 sends a new INVITE request to SIP gateway 1.
F12	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends an ACK to SIP gateway 2. The ACK confirms that SIP gateway 1 has received the 200 OK response from SIP gateway 2.
F13	BYE—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a BYE request to SIP gateway 2. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
F14	200 OK—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.

SIP Call Using RFC2833 for DTMF-Relay Output from GW1 Side

Call Trace with Different Payload Types:

In this trace both ends have been configured to use NTE dtmf relay mode. However the payload types for NTE packets have different values on both ends. The originator is configured to use value 115, terminator will use default value of 101. The payload value can be configured at the dial peer with **rtp payload-type nte 101** command. Because 101 is default payload type and this value is not being used at the Originator for any other payload type, the two ends agree to use payload type 101.

```

GW1 ---INVITE---> GW2 (F1)
GW1 <-100Trying-- GW2 (F2)
GW1 <-183SesProg- GW2 (F3)
GW1 <---200OK---- GW2 (F4)
GW1 -----ACK-----> GW2 (F5)
GW1 <-----BYE----- GW2 (F6)
GW1 ---200OK-----> GW2 (F7)

```

(F1)Sent:

```

INVITE sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:36602@172.18.193.120;user=phone>
Date: Mon, 01 Mar 1993 00:23:13 GMT
Call-ID: C6310C75-14F111CC-801D9792-19DC655A@172.18.193.98
Cisco-Guid: 3301622965-351343052-2149291922-433874266
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 730945393
Contact: <sip:36601@172.18.193.98:5060;user=phone>
Expires: 180
Content-Type: application/sdp
Content-Length: 216

```

```

v=0
o=CiscoSystemsSIP-GW-UserAgent 6351 4261 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 18720 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

```

**Note**

The INVITE is sent, and it requests that RFC 2833 be used for DTMF-Relay. This is communicated in the SDP body, in the m= and a= lines.

```
(F2)
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:36602@172.18.193.120;user=phone>
Date: Mon, 01 Mar 1993 00:23:14 GMT
Call-ID: C6310C75-14F111CC-801D9792-19DC655A@172.18.193.98
Timestamp: 730945393
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Content-Length: 0
```

```
(F3)
Received:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:36602@172.18.193.120;user=phone>
Date: Mon, 01 Mar 1993 00:23:14 GMT
Call-ID: C6310C75-14F111CC-801D9792-19DC655A@172.18.193.98
Timestamp: 730945393
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Content-Type: application/sdp
Session: Media
Content-Length: 217
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 637 6719 IN IP4 172.18.193.120
s=SIP Call
c=IN IP4 172.18.193.120
t=0 0
m=audio 18210 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

**Note**

The Terminating GW responds back that it can support RFC 2833 for DTMF-Relay. It communicates this in its SDP body, in the m= and a= lines.

```
(F4)
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:36602@172.18.193.120;user=phone>;tag=154C54-A4F
Date: Mon, 01 Mar 1993 00:23:14 GMT
Call-ID: C6310C75-14F111CC-801D9792-19DC655A@172.18.193.98
Timestamp: 730945393
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:36602@172.18.193.120:5060;user=phone>
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 217

v=0
o=CiscoSystemsSIP-GW-UserAgent 637 6719 IN IP4 172.18.193.120
s=SIP Call
c=IN IP4 172.18.193.120
t=0 0
m=audio 18210 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
(F5)
Sent:
ACK sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>
To: <sip:36602@172.18.193.120;user=phone>;tag=154C54-A4F
Date: Mon, 01 Mar 1993 00:23:13 GMT
Call-ID: C6310C75-14F111CC-801D9792-19DC655A@172.18.193.98
Max-Forwards: 6
Content-Length: 0
CSeq: 101 ACK
```

```
(F6)
Received:
BYE sip:36601@172.18.193.98:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.120:5060
From: <sip:36602@172.18.193.120;user=phone>;tag=154C54-A4F
To: "36601"<sip:36601@172.18.193.98>
Date: Mon, 01 Mar 1993 00:23:15 GMT
Call-ID: C6310C75-14F111CC-801D9792-19DC655A@172.18.193.98
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 730945400
CSeq: 101 BYE
Content-Length: 0
```

```
(F7)
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.120:5060
```

```

From: <sip:36602@172.18.193.120;user=phone>;tag=154C54-A4F
To: "36601"<sip:36601@172.18.193.98>
Date: Mon, 01 Mar 1993 00:23:19 GMT
Call-ID: C6310C75-14F111CC-801D9792-19DC655A@172.18.193.98
Server: Cisco-SIPGateway/IOS-12.x
Timestamp: 730945400
Content-Length: 0
CSeq: 101 BYE

```

FigureC-3 SIP Call Using RFC2833 for DTMF-Relay

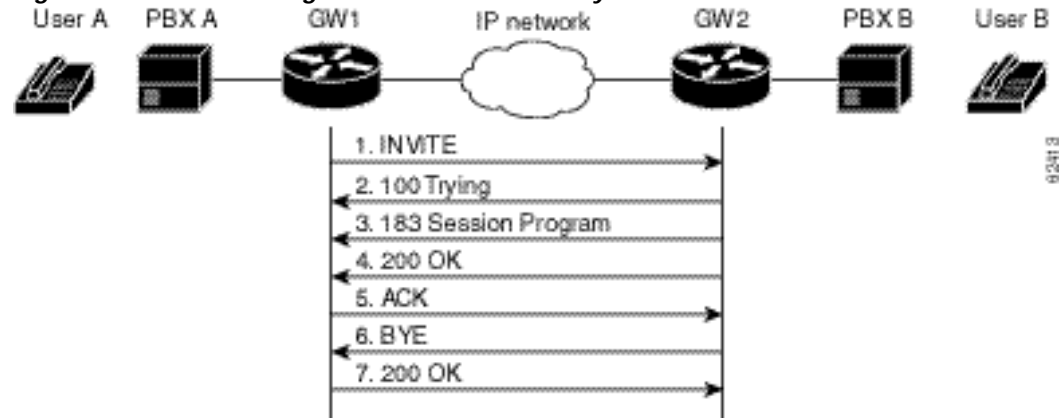


Table C-3 SIP Call Using RFC 2833 for DTMF-Relay Output from GW1 Side

Step	Action	Description
F1	INVITE—SIP Gateway 1 to SIP Gateway 2	<p>SIP gateway 1 sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an e-mail address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). In this example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:3111100@64.102.17.80:5060; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a username. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability (designated by m=) User A is ready to receive is specified. In this case m=audio 17158 RTP/AVP 0 18. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
F2	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
F3	183 Session Progress/QoS—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 sends a 183 Session Progress response to SIP gateway 1. The 183 Session Progress response indicates that SIP gateway 2 has located, and is trying to perform inband alerting for the caller.</p> <p>Additionally Quality of Service (QoS) reserves sufficient network-layer resources to guarantee bandwidth and bounds on packet loss, delay, and jitter; thus ensuring that the called party's phone rings only after bandwidth required for the call has been successfully reserved.</p>
F4	200 OK—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made.</p> <p>If User B supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.</p>
F5	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a an ACK to SIP gateway 2. The ACK confirms that SIP gateway 1 has received the 200 OK response from SIP gateway 2.

Step	Action	Description
F6	BYE—SIP Gateway 2 to SIP Gateway 1	SIP Gateway 2 sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
F7	200 OK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 1 that SIP gateway 2 has received the BYE request.

SIP Provisional Response Output from GW1 Side

```

GW1 ---INVITE---> GW2 (F1)
GW1 <-100Trying-- GW2 (F2)
GW1 <-183SesProg- GW2 (F3)
GW1 ---PRACK----> GW2 (F4)
GW1 <-200/PRACK-- GW2 (F5)
GW1 <---200OK---- GW2 (F6)
GW1 -----ACK-----> GW2 (F7)
GW1 <-----BYE----- GW2 (F8)
GW1 ---200OK-----> GW2 (F9)

```



Note

The call-flows and headers are per IETF draft 'draft-ietf-sip-100rel-03.txt'

```

(F1)
Sent:
INVITE sip:36602@172.18.193.120;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601" <sip:36601@172.18.193.98>;tag=27900-D65
To: <sip:36602@172.18.193.120;user=phone>
Date: Mon, 01 Mar 1993 00:02:42 GMT
Call-ID: E85BBD00-14EE11CC-800B9D18-7AB2874B@172.18.193.98
Supported: 100rel
Cisco-Guid: 3878326272-351146444-2148113688-2058520395
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 730944162
Contact: <sip:36601@172.18.193.98:5060;user=phone>
Expires: 180
Content-Type: application/sdp
Content-Length: 183

v=0
o=CiscoSystemsSIP-GW-UserAgent 531 2364 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 18582 RTP/AVP 0 18
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000

```

**Note**

The SIP INVITE is sent requesting support for Provisional Response messages. This is communicated in the Supported: header.

```
(F2)
Received:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=27900-D65
To: <sip:36602@172.18.193.120;user=phone>;tag=20B28-E2A
Date: Mon, 01 Mar 1993 00:02:13 GMT
Call-ID: E85BBD00-14EE11CC-800B9D18-7AB2874B@172.18.193.98
Timestamp: 730944162
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Content-Length: 0
```

```
(F3)
Received:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=27900-D65
To: <sip:36602@172.18.193.120;user=phone>;tag=20B28-E2A
Date: Mon, 01 Mar 1993 00:02:13 GMT
Call-ID: E85BBD00-14EE11CC-800B9D18-7AB2874B@172.18.193.98
Timestamp: 730944162
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:36602@172.18.193.120:5060;user=phone>
CSeq: 101 INVITE
Require: 100rel
RSeq: 8289
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 162
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 8320 2989 IN IP4 172.18.193.120
s=SIP Call
c=IN IP4 172.18.193.120
t=0 0
m=audio 18036 RTP/AVP 18
a=rtpmap:18 G729/8000
```

**Note**

The terminating GW responds back that it supports Provisional Responses, but sending the Require: and Content-Disposition: header.

```
(F4)
Sent:
PRACK sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=27900-D65
To: <sip:36602@172.18.193.120;user=phone>;tag=20B28-E2A
Date: Mon, 01 Mar 1993 00:02:42 GMT
Call-ID: E85BBD00-14EE11CC-800B9D18-7AB2874B@172.18.193.98
CSeq: 102 PRACK
RAck: 8289 101 INVITE
Content-Length: 0
```



Note The originating gateway responds back to the Provisional Response with a PRACK message.

```
(F5)
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=27900-D65
To: <sip:36602@172.18.193.120;user=phone>;tag=20B28-E2A
Date: Mon, 01 Mar 1993 00:02:13 GMT
Call-ID: E85BBD00-14EE11CC-800B9D18-7AB2874B@172.18.193.98
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 102 PRACK
Content-Length: 0
```

```
(F6)
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=27900-D65
To: <sip:36602@172.18.193.120;user=phone>;tag=20B28-E2A
Date: Mon, 01 Mar 1993 00:02:13 GMT
Call-ID: E85BBD00-14EE11CC-800B9D18-7AB2874B@172.18.193.98
Timestamp: 730944162
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:36602@172.18.193.120:5060;user=phone>
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 162
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 8320 2989 IN IP4 172.18.193.120
s=SIP Call
c=IN IP4 172.18.193.120
t=0 0
m=audio 18036 RTP/AVP 18
a=rtpmap:18 G729/8000
```

```
(F7)
Sent:
ACK sip:36602@172.18.193.120:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.98:5060
From: "36601"<sip:36601@172.18.193.98>;tag=27900-D65
To: <sip:36602@172.18.193.120;user=phone>;tag=20B28-E2A
Date: Mon, 01 Mar 1993 00:02:42 GMT
Call-ID: E85BBD00-14EE11CC-800B9D18-7AB2874B@172.18.193.98
Max-Forwards: 6
Content-Length: 0
CSeq: 101 ACK
```

```
(F8)
Received:
BYE sip:36601@172.18.193.98:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.120:5060
From: <sip:36602@172.18.193.120;user=phone>;tag=20B28-E2A
To: "36601"<sip:36601@172.18.193.98>;tag=27900-D65
Date: Mon, 01 Mar 1993 00:02:28 GMT
Call-ID: E85BBD00-14EE11CC-800B9D18-7AB2874B@172.18.193.98
```

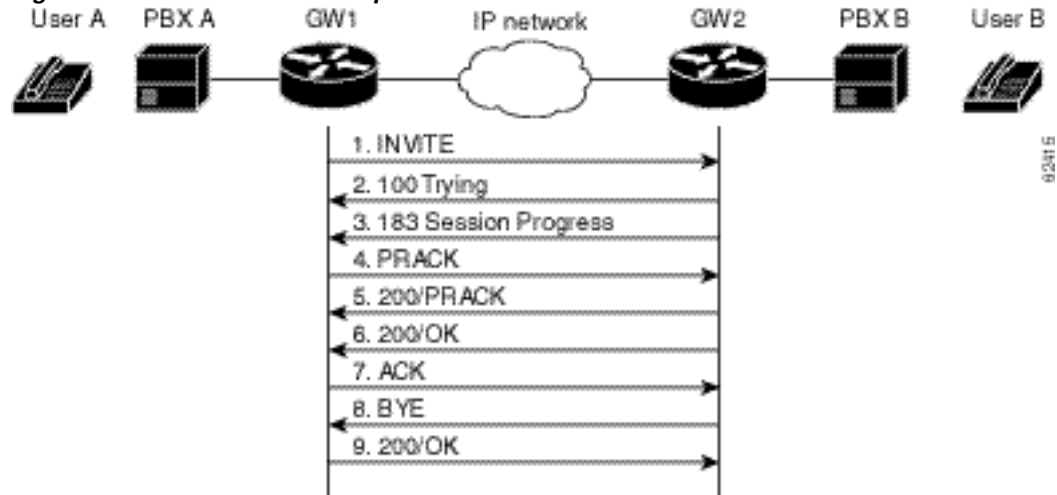
```

User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 730944154
CSeq: 101 BYE
Content-Length: 0

(F9)
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.120:5060
From: <sip:36602@172.18.193.120;user=phone>;tag=20B28-E2A
To: "36601"<sip:36601@172.18.193.98>;tag=27900-D65
Date: Mon, 01 Mar 1993 00:03:03 GMT
Call-ID: E85BBD00-14EE11CC-800B9D18-7AB2874B@172.18.193.98
Server: Cisco-SIPGateway/IOS-12.x
Timestamp: 730944154
Content-Length: 0
CSeq: 101 BYE

```

Figure C-4 SIP Provisional Response



TableC-4 SIP Provisional Response Output from GW1 Side

Step	Action	Description
F1	INVITE—SIP Gateway 1 to SIP Gateway 2	<p>SIP gateway 1 sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an e-mail address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). In this example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:3111100@64.102.17.80:5060; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a user name. • PBX A is identified as the call session initiator in the From field. • A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. • The transaction number within a single call leg is identified in the CSeq field. • The media capability (designated by m=) User A is ready to receive is specified. In this case m=audio 17158 RTP/AVP 0 18. • The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
F2	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
F3	183 Session Progress/QoS—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 sends a 183 Session Progress response to SIP gateway 1. The 183 Session Progress response indicates that SIP gateway 2 has located, and is trying to perform inband alerting for the caller.</p> <p>Additionally Quality of Service (QoS) reserves sufficient network-layer resources to guarantee bandwidth and bounds on packet loss, delay, and jitter; thus ensuring that the called party's phone rings only after bandwidth required for the call has been successfully reserved.</p>
F4	PRACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 acknowledges the 183 Session Progress response with a PRovisional ACKnowledgement (PRACK).
F5	200/PRACK—SIP Gateway 2 to SIP Gateway 1	<p>The 200OK/PRovisional ACKnowledgement (PRACK) response notifies SIP gateway 1 that the connection has been made.</p> <p>The SIP gateway generates this response when the PBX indicates that the connection has been made. On receiving this response, the gateway forwards the response to the corresponding party and responds with a PRACK.</p>

Step	Action	Description
F6	200 OK—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made. If User B supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400 Bad Request response with a 304 Warning header field.
F7	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a an ACK to SIP gateway 2. The ACK confirms that SIP gateway 1 has received the 200 OK response from SIP gateway 2.
F8	BYE—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a BYE request to SIP gateway 1. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
F9	200 OK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 1 that SIP gateway 2 has received the BYE request.

SIP T.38 Call with QoS Enabled Output

```

GW1 -----INV-----> GW2 (F1)
GW1 <--100 Trying- GW2 (F2)
GW1 <--183 QoS --- GW2 (F3)
GW1 -----PRACK-----> GW2 (F4)
GW1 <--200/PRACK-- GW2 (F5)
GW1 -----COMET-----> GW2 (F6)
GW1 <--200/COMET-- GW2 (F7)
GW1 <--183SesProg- GW2 (F8)
GW1 -----PRACK-----> GW2 (F9)
GW1 <--200/PRACK-- GW2 (F10)
GW1 <---200OK----- GW2 (F11)
GW1 -----ACK-----> GW2 (F12)
GW1 <-reINVITE/FAX GW2 (F13)
GW1 -----200OK-----> GW2 (F14)
GW1 <---ACK----- GW2 (F15)
GW1 -----BYE-----> GW2 (F16)
GW1 <---200OK----- GW2 (F17)

```

**Note**

T.38 support is per T.38 Annex-D and the IETF draft 'draft-mule-sip-t38callflows-01.txt'. The call-flow below shows a call between two SIP Gateways, which are connected (via TDM) to fax machines.

```
(F1)
Received:
INVITE sip:2000@172.18.193.135;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.187:5060;branch=1
Via: SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Supported: 100rel
Cisco-Guid: 4143344000-1203638741-2153637452-3172295218
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 5
Timestamp: 989858562
Contact: <sip:1000@172.18.193.196:5060;user=phone>
Expires: 180
Content-Type: application/sdp
Content-Length: 244

v=0
o=CiscoSystemsSIP-GW-UserAgent 5535 1304 IN IP4 172.18.193.196
s=SIP Call
c=IN IP4 172.18.193.196
t=0 0
m=audio 16608 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=qos:optional sendrecv
```

**Note**

Initial call is received. QoS is requested.

```
(F2)
Sent:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.18.193.187:5060;branch=1,SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Timestamp: 989858562
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Content-Length: 0
```

```
(F3)
Sent:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.18.193.187:5060;branch=1,SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Timestamp: 989858562
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:2000@172.18.193.135:5060;user=phone>
CSeq: 101 INVITE
Require: 100rel
RSeq: 5700
Content-Type: application/sdp
Content-Disposition: precondition;handling=required
Content-Length: 247
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 5201 1828 IN IP4 172.18.193.135
s=SIP Call
c=IN IP4 172.18.193.135
t=0 0
m=audio 18036 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101
a=qos:optional sendrecv confirm
```

**Note**

QoS is sent.

```
(F4)
Received:
PRACK sip:2000@172.18.193.135:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
CSeq: 102 PRACK
RAck: 5700 101 INVITE
Content-Length: 0
```

```
(F5)
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 102 PRACK
Content-Length: 0
```

```
(F6)
Received:
COMET sip:2000@172.18.193.135:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
CSeq: 103 COMET
Content-Type: application/sdp
Content-Length: 243
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 5535 1304 IN IP4 172.18.193.196
s=SIP Call
c=IN IP4 172.18.193.196
t=0 0
m=audio 16608 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=qos:succes sendrecv
```



Note Bidirectional QoS has been confirmed.

```
(F7)
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 103 COMET
Content-Length: 0
```

```
(F8)
Sent:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.18.193.187:5060;branch=1,SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Timestamp: 989858562
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:2000@172.18.193.135:5060;user=phone>
CSeq: 101 INVITE
Require: 100rel
RSeq: 5701
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 214
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 5201 1828 IN IP4 172.18.193.135
s=SIP Call
c=IN IP4 172.18.193.135
t=0 0
m=audio 18036 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101
```

```
(F9)
Received:
PRACK sip:2000@172.18.193.135:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
CSeq: 104 PRACK
RAck: 5701 101 INVITE
Content-Length: 0
```

```
(F10)
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.196:5060
```

```
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 104 PRACK
Content-Length: 0
```

```
(F11)
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.187:5060;branch=1,SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Timestamp: 989858562
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:2000@172.18.193.135:5060;user=phone>
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 214
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 5201 1828 IN IP4 172.18.193.135
s=SIP Call
c=IN IP4 172.18.193.135
t=0 0
m=audio 18036 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101
```

```
(F12)
Received:
ACK sip:2000@172.18.193.135:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:42:42 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Max-Forwards: 6
Content-Length: 0
CSeq: 101 ACK
```

```
(F13)
Sent:
INVITE sip:1000@172.18.193.196:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.135:5060
From: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
To: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
Date: Mon, 14 May 2001 17:43:11 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Supported: 100rel
Cisco-Guid: 4143344000-1203638741-2153637452-3172295218
User-Agent: Cisco-SIPGateway/IOS-12.x
CSeq: 101 INVITE
Max-Forwards: 6
Timestamp: 989858591
Contact: <sip:2000@172.18.193.135:5060;user=phone>
Expires: 180
```

```
Content-Type: application/sdp
Content-Length: 403
```

```
v=0
o=CiscoSystemsSIP-GW-UserAgent 5201 1829 IN IP4 172.18.193.135
s=SIP Call
c=IN IP4 172.18.193.135
t=0 0
m=image 18036 udptl t38
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:72
a=T38FaxUdpEC:t38UDPRedundancy
a=qos:optional sendrecv
```

**Note**

T.38 is signaled via a reINVITE message with the T.38 parameters defined in the SDP body.

```
(F14)
Received:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.135:5060
From: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
To: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
Date: Mon, 14 May 2001 17:43:11 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Server: Cisco-SIPGateway/IOS-12.x
Contact: <sip:1000@172.18.193.196:5060;user=phone>
CSeq: 101 INVITE
Content-Type: application/sdp
Content-Length: 138

v=0
o=CiscoSystemsSIP-GW-UserAgent 5535 1305 IN IP4 172.18.193.196
s=SIP Call
c=IN IP4 172.18.193.196
t=0 0
m=image 16608 udpt1 t38
```

T.38 is acknowledged. At this point the FAX will be sent using UDPTL.

```
(F15)
Sent:
ACK sip:1000@172.18.193.196:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.135:5060
From: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
To: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
Date: Mon, 14 May 2001 17:43:11 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
Max-Forwards: 6
Content-Length: 0
CSeq: 101 ACK
```

```
(F16)
Received:
BYE sip:2000@172.18.193.135:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:43:11 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 989858617
CSeq: 105 BYE
Content-Length: 0
```

```
(F17)
Sent:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.18.193.196:5060
From: "1000"<sip:1000@172.18.193.196>;tag=14B99A90-269E
To: <sip:2000@172.18.193.187;user=phone>;tag=14B968AC-2668
Date: Mon, 14 May 2001 17:43:37 GMT
Call-ID: F8C02D00-47BE11D5-805FE64C-BD156232@172.18.193.196
```

```

Server: Cisco-SIPGateway/IOS-12.x
Timestamp: 989858617
Content-Length: 0
CSeq: 105 BYE

```

FigureC-5 SIP T.38 Call with QoS Enabled

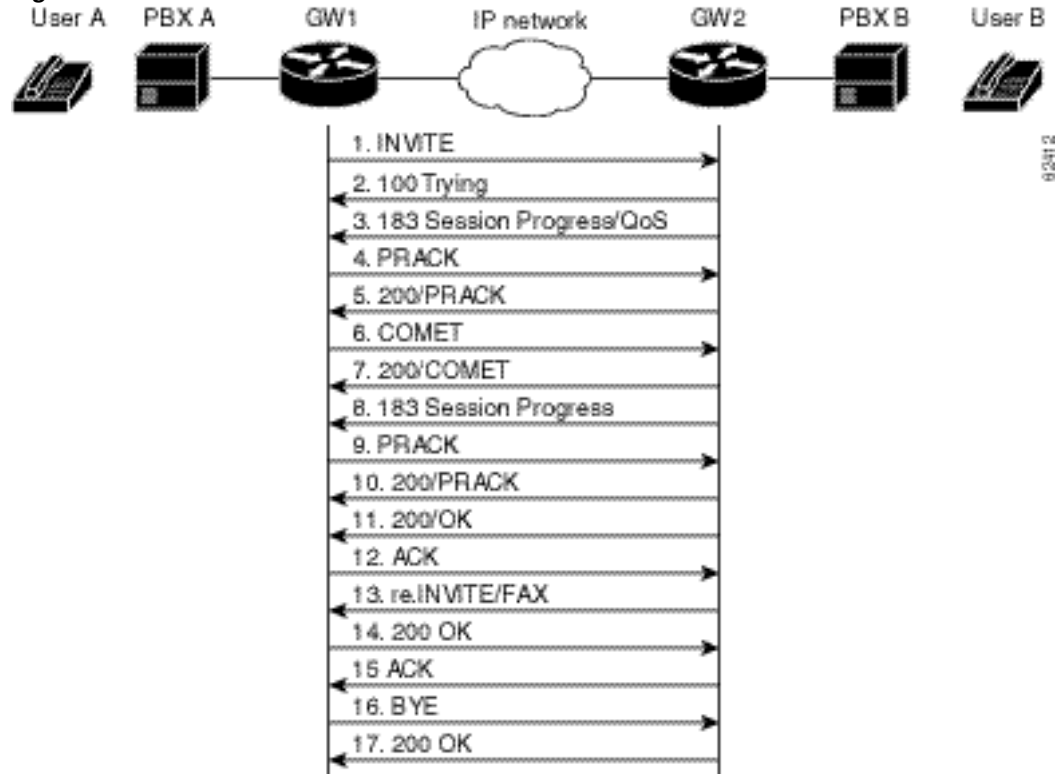


Table C-5 SIP T.38 Call with QoS Enabled Output

Step	Action	Description
F1	INVITE—SIP Gateway 1 to SIP Gateway 2	<p>SIP gateway 1 sends an INVITE request to SIP gateway 2. The INVITE request is an invitation to User B to participate in a call session.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> The phone number of User B is inserted in the Request-URI field in the form of a SIP URL. The SIP URL identifies the address of User B and takes a form similar to an e-mail address (<i>user@host</i> where <i>user</i> is the telephone number and <i>host</i> is either a domain name or a numeric network address). In this example, the Request-URI field in the INVITE request to User B appears as “INVITE sip:3111100@64.102.17.80:5060; user=phone.” The “user=phone” parameter distinguishes that the Request-URI address is a telephone number rather than a username. PBX A is identified as the call session initiator in the From field. A unique numeric identifier is assigned to the call and is inserted in the Call-ID field. The transaction number within a single call leg is identified in the CSeq field. The media capability (designated by m=) User A is ready to receive is specified. In this case m=audio 17158 RTP/AVP 0 18. The port on which SIP gateway 1 is prepared to receive the RTP data is specified.
F2	100 Trying—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 100 Trying response to the INVITE request sent by SIP gateway 1. The 100 Trying response indicates that the INVITE request has been received by SIP gateway 2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
F3	183 Session Progress/QoS—SIP Gateway 2 to SIP Gateway 1	<p>SIP gateway 2 sends a 183 Session Progress response to SIP gateway 1. The 183 Session Progress response indicates that SIP gateway 2 has located, and is trying to perform inband alerting for the caller.</p> <p>Additionally Quality of Service (QoS) reserves sufficient network-layer resources to guarantee bandwidth and bounds on packet loss, delay, and jitter; thus ensuring that the called party's phone rings only after bandwidth required for the call has been successfully reserved.</p>
F4	PRACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 acknowledges the 183 Session Progress response with a PRovisional ACKnowledgement (PRACK).
F5	200/PRACK—SIP Gateway 2 to SIP Gateway 1	<p>The 200OK/PRovisional ACKnowledgement (PRACK) response notifies SIP gateway 1 that the connection has been made.</p> <p>The SIP gateway generates this response when the PBX indicates that the connection has been made. On receiving this response, the gateway forwards the response to the corresponding party and responds with a PRACK.</p>
F6	COMET—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a COnditions MET (COMET) response to SIP gateway 2 indicating that the pre-conditions have been met.
F7	200/COMET—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a COnditions MET (COMET) response to SIP gateway 1 indicating that the preconditions have been met.

Step	Action	Description
F8	183 Session Progress— SIP Gateway 2 to SIP Gateway 1	The 183 Session Progress response is used to perform inband alerting for the caller.
F9	PRACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 acknowledges the 183 Session Progress response with a PRovisional ACKnowledgement (PRACK).
F10	200/PRACK—SIP Gateway 2 to SIP Gateway 1	The 200OK/PRovisional ACKnowledgement (PRACK) response notifies SIP gateway 1 that the connection has been made. The SIP gateway generates this response when the PBX indicates that the connection has been made. On receiving this response, the gateway forwards the response to the corresponding party and responds with a PRACK.
F11	200 OK—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 1 that the connection has been made. If User B supports the media capability advertised in the INVITE message sent by SIP gateway 1, it advertises the intersection of its own and User A's media capability in the 200 OK response. If User B does not support the media capability advertised by User A, it sends back a 400Bad Request response with a 304 Warning header field.
F12	ACK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a an ACK to SIP gateway 2. The ACK confirms that SIP gateway 1 has received the 200 OK response from SIP gateway 2.
F13	reINVITE/FAX— SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 receives the initial Fax Tones (T4/T30). It sends a SIP reINVITE to negotiate the T.38 Fax-Relay capabilities.
F14	200 OK—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a 200 OK response to SIP gateway 2. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.
F15	ACK—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends an ACK to SIP gateway 1. The ACK confirms that SIP gateway 2 has received the 200 OK response from SIP gateway 1.
F16	BYE—SIP Gateway 1 to SIP Gateway 2	SIP gateway 1 sends a BYE request to SIP gateway 2. The BYE request indicates that User B wants to release the call. Because it is User B that wants to terminate the call, the Request-URI field is now replaced with PBX A's SIP URL and the From field contains User B's SIP URL.
F17	200 OK—SIP Gateway 2 to SIP Gateway 1	SIP gateway 2 sends a 200 OK response to SIP gateway 1. The 200 OK response notifies SIP gateway 2 that SIP gateway 1 has received the BYE request.