



Session Initiation Protocol Gateway Call Flows

This document provides information about the Session Initiation Protocol (SIP) call flows as they are implemented in the Cisco IOS Release 12.1(1)T and used in Cisco Voice over IP gateways. The primary focus of this document is the unique SIP methods exchanged during the call flows.

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SIP Messages Overview

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.

SIP uses six request methods:

- 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. Gateways do not support the REGISTER method.

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

- SIP 1xx—Informational Responses
- SIP 2xx—Successful Responses

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- SIP 3xx—Redirection Responses
- SIP 4xx—Client Failure Responses
- SIP 5xx—Server Failure Responses
- SIP 6xx—Global Failure Responses

Successful Call Flow Scenarios

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

- Gateway-to-Gateway Call, page 2
- Gateway-to-Gateway Call via Redirect Server, page 5
- Gateway-to-Gateway Call via a Proxy Server, page 7
- Gateway-to-IP Desktop Call, page 11

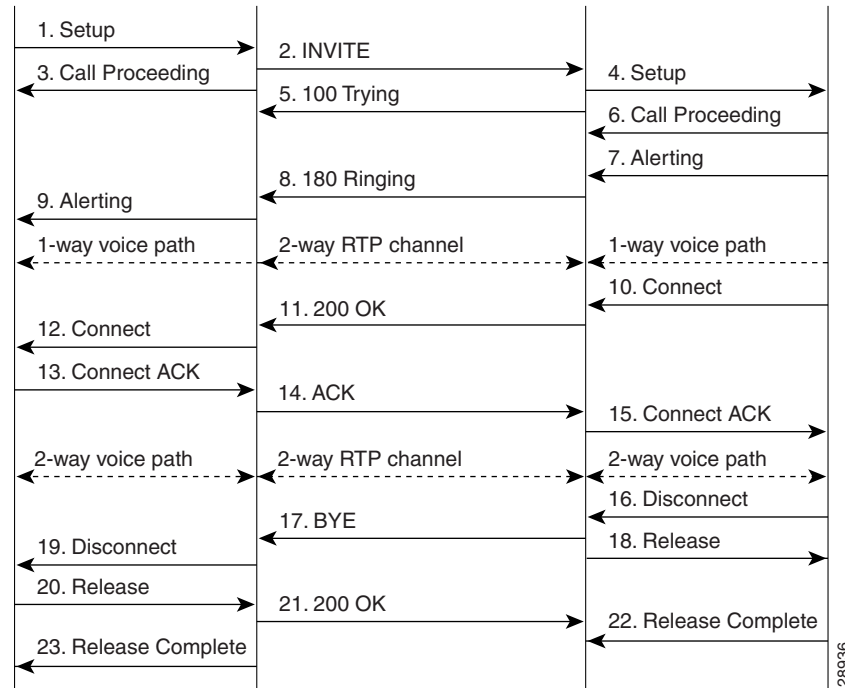
Gateway-to-Gateway Call

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 GW1 (SIP Gateway) via a T1/E1. User B is located at PBX B. PBX B is connected to GW2 (SIP Gateway) via a T1/E1. User B's phone number is 555-0002. GW1 is connected to GW2 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 Gateway-to-Gateway Call—Successful Setup and Disconnect



Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> GW2	GW1 sends a SIP INVITE request to GW2. 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. 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 GW1 is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	Setup—GW2 -> PBX B	GW2 receives the INVITE request from GW1 and initiates a Call Setup with User B via PBX B.

Successful Call Flow Scenarios

Step	Action	Description
5	100 Trying—GW2 -> GW1	GW2 sends a SIP 100 Trying response to the INVITE request sent by GW1. The 100 Trying response indicates that the INVITE request has been received by GW2 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 -> GW2	PBX B sends a Call Proceeding message to GW2 to acknowledge the Call Setup request.
7	Alerting—PBX B -> GW2	PBX B locates User B and sends an Alert message to GW2. User B's phone begins ringing.
8	180 Ringing—GW2 -> GW1	GW2 sends a SIP 180 Ringing response to GW1. The 180 Ringing response indicates that GW2 has located, and is trying to alert, User B.
9	Alerting—GW1 -> PBX A	GW1 sends an Alert message to User A via PBX A. The Alert message indicates that GW1 has received a 180 Ringing response from GW2. User A hears the ringback tone that indicates that User B is being alerted.
10	Connect—PBX B -> GW2	User B answers phone. PBX B sends a Connect message to GW2. The Connect message notifies GW2 that the connection has been made.
11	200 OK—GW1 -> GW2	GW2 sends a SIP 200 OK response to GW1. The 200 OK response notifies GW1 that the connection has been made. If User B 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 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.
12	Connect—GW1 -> PBX A	GW1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
13	Connect ACK—PBX A -> GW1	PBX A acknowledges GW1's Connect message.
14	ACK—GW1 and GW2	GW1 sends a SIP ACK request to GW2. The ACK request confirms that User A has received the 200 OK response from User B. The ACK request might contain a message body with the final session description to be used by User B. If the message body of the ACK request is empty, User B uses the session description in the INVITE request.
15	Connect ACK—GW2 -> PBX B	GW2 acknowledges PBX B's Connect message. The call session is now active over a two-way voice path via Real Time Transport Protocol (RTP).
16	Disconnect—PBX B -> GW2	Once User B hangs up, PBX B sends a Disconnect message to GW2. The Disconnect message starts the call session termination process.
17	BYE F11—GW2 -> GW1	GW2 sends a SIP BYE request to GW1. 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—GW2 -> PBX B	GW2 sends a Release message to PBX B.
19	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
20	Release—PBX A -> GW1	PBX A sends a Disconnect message to GW1.
21	200 OK F12—GW1 -> GW2	GW1 sends a SIP 200 OK response to GW2. The 200 OK response notifies GW2 that GW1 has received the BYE request.
22	Release Complete—PBX B -> GW2	PBX B sends a Release Complete message to GW2.
23	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the session is terminated.

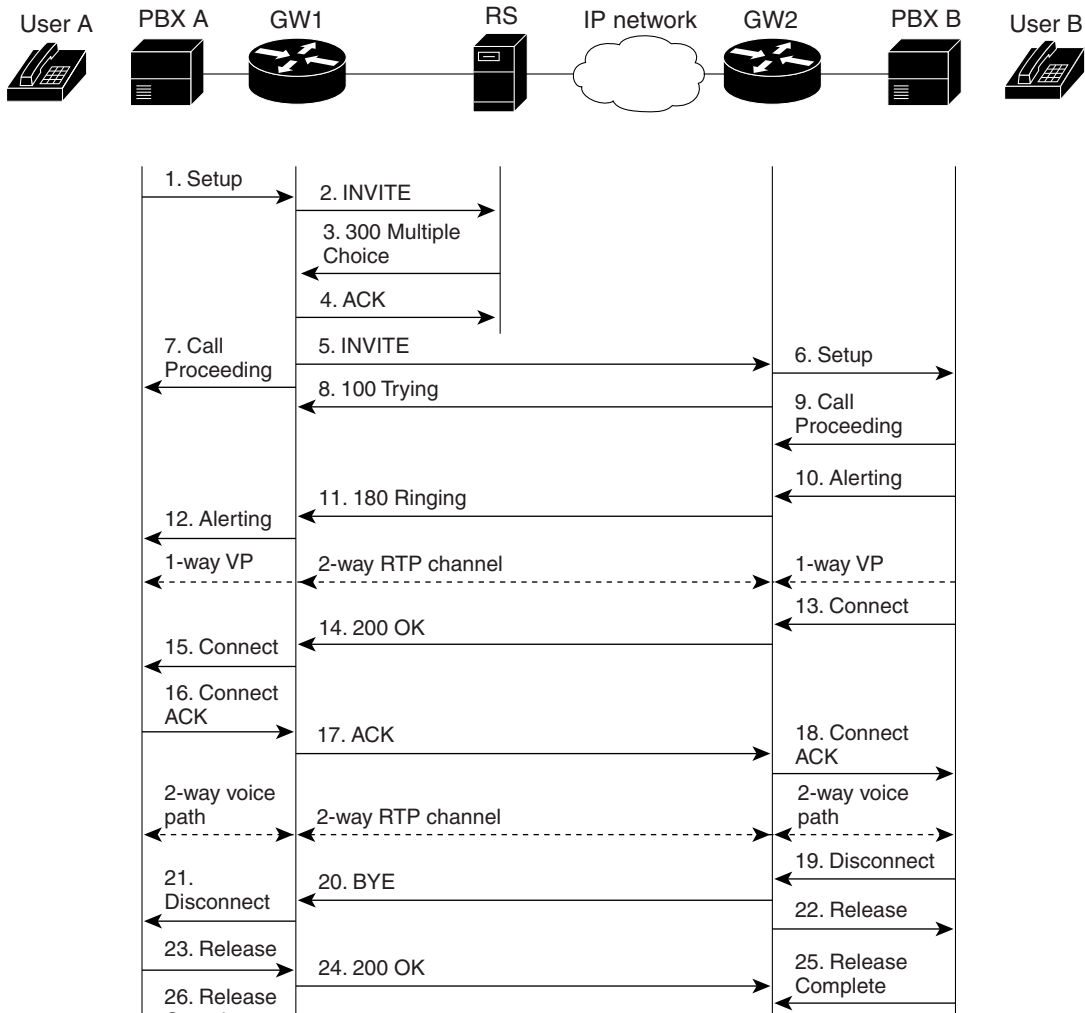
Gateway-to-Gateway Call via Redirect Server

Figure 2 illustrates a successful gateway-to-gateway call setup and disconnect via a 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 GW1 (SIP Gateway) via a T1/E1. GW1 is using a redirect server. User B is located at PBX B. PBX B is connected to GW2 (SIP Gateway) via a T1/E1. User B’s phone number is 555-0002. GW1 is connected to GW2 over an IP network.

The call flow scenario is as follows:

- 1 User A calls User B via GW1 using a redirect server.
- 2 User B answers the call.
- 3 User B hangs up.

Figure 2 Gateway-to-Gateway Call via a Redirect Server—Successful Setup and Disconnect



Successful Call Flow Scenarios

Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> RS	<p>GW1 sends a SIP INVITE request to the 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 GW1 is prepared to receive the RTP data is specified.
3	300 Multiple Choice—RS -> GW1	The redirect server sends GW1 a SIP 300 Multiple Choice response. The 300 Multiple Choice response indicates that the 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 redirect server returns these possible addresses to GW1 in the 300 Multiple Choice response.
4	ACK—GW1 -> RS	GW1 acknowledges the 300 Multiple Choice response with an ACK request.
5	INVITE—GW1 -> GW2	GW1 sends a new INVITE request to GW2. 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—GW2 -> PBX B	GW2 receives the INVITE request from GW1 and initiates a Call Setup with User B via PBX B.
7	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
8	100 Trying—GW2 -> GW1	GW2 sends a SIP 100 Trying response to the INVITE request sent by GW1. The 100 Trying response indicates that the INVITE request has been received by GW2 but that User B has not yet been located.
9	180 Ringing—GW2 -> GW1	GW2 sends a SIP 180 Ringing response to GW1. The 180 Ringing response indicates that GW2 has located, and is trying to alert, User B.
10	Alerting—GW1 -> PBX A	GW1 sends an Alert message to PBX A. User A hears ringback tone.
11	Connect—PBX B to GW2	User B answers phone. PBX B sends a Connect message to GW2. The Connect message notifies GW2 that the connection has been made.
12	200 OK—GW2 -> GW1	<p>GW2 sends a 200 OK response to GW1. The 200 OK response notifies GW1 that the connection has been made.</p> <p>If User B 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 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>
13	Connect—GW1 -> PBX A	GW1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
14	Connect ACK—PBX A -> GW1	PBX A acknowledges GW1’s Connect message.

Step	Action	Description
15	ACK—GW1 -> GW2	GW1 sends a SIP ACK request to GW2. The ACK request confirms that the 200 OK response has been received. The ACK request might contain a message body with the final session description to be used by User B. If the message body of the ACK request is empty, User B uses the session description in the INVITE request. The call is now in progress over a two-way voice path via RTP.
16	Connect ACK—GW2 -> PBX B	GW2 acknowledges PBX B's Connect message.
17	Disconnect—PBX B -> GW2	Once User B hangs up, PBX B sends a Disconnect message to GW2. The Disconnect message starts the call session termination process.
18	BYE—GW2 -> GW1	GW2 sends a SIP BYE request to GW1. 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.
19	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
20	Release—GW2 -> PBX B	GW2 sends a Release message to PBX B.
21	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
22	200 OK—GW1 -> GW2	GW1 sends a 200 OK response to GW2. The 200 OK response notifies GW2 that GW1 has received the BYE request.
23	Release Complete—PBX B -> GW2	PBX B sends a Release Complete message to GW2.
24	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the session is terminated.

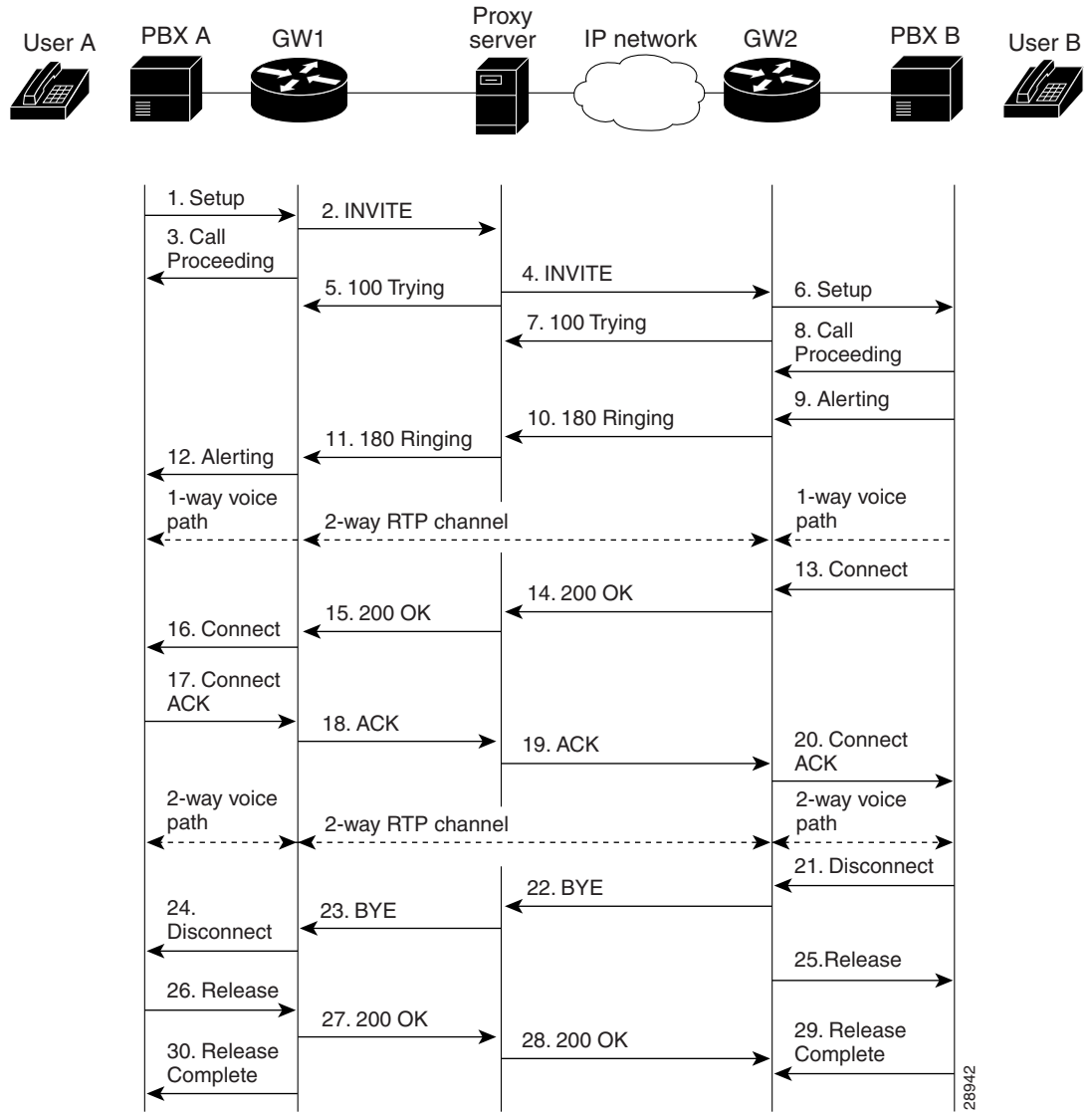
Gateway-to-Gateway Call via a Proxy Server

Figure 3 illustrates a successful gateway-to-gateway call setup and disconnect via a proxy server. 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 GW1 (SIP Gateway) via a T1/E1. GW1 is using a proxy server. GW1 is connected to GW2 over an IP network. User B is located at PBX B. PBX B is connected to GW2 (a SIP Gateway) via a T1/E1. User B's phone number is 555-0002.

The call flow is as follows:

- 1 User A calls User B via GW1 using a proxy server.
- 2 User B answers the call.
- 3 User B hangs up.

Figure 3 Gateway-to-Gateway Call via a Proxy Server—Successful Setup and Disconnect



Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—GW1 -> PS	<p>GW1 sends a SIP INVITE request to the 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 GW1 is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	INVITE—PS -> GW2	The 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 GW1, 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 GW2.
5	100 Trying—PS -> GW1	The proxy server sends a SIP 100 Trying response to GW1.
6	Setup—GW2 -> PBX B	GW2 receives the INVITE request from the proxy server and initiates a Call Setup with User B via PBX B.
7	100 Trying—GW2 -> PS	GW2 sends a SIP 100 Trying response to the proxy server. The proxy server might or might not forward the 100 Trying response to GW1.
8	Call Proceeding—PBX B -> GW2	PBX B sends a Call Proceeding message to GW2 to acknowledge the Call Setup request.
9	Alerting—PBX B -> GW2	PBX B locates User B and sends an Alert message to GW2. User B’s phone begin to ring.
10	180 Ringing—GW2 -> PS	GW2 sends a SIP 180 Ringing response to the proxy server.
11	180 Ringing—PS -> GW1	The proxy server forwards the 180 Ringing response to GW1.
12	Alerting—GW1 -> PBX A	GW1 sends an Alert message to User A via PBX A. The Alert message indicates that GW1 has received a 180 Ringing response. User A hears the ringback tone that indicates that User B is being alerted.
13	Connect—PBX B -> GW2	User B answers the phone. PBX B sends a Connect message to GW2. The connect message notifies GW2 that the connection has been made.
14	200 OK—GW2 -> PS	<p>GW2 sends a SIP 200 OK response to the proxy server. The 200 OK response notifies the proxy server that the connection has been made.</p> <p>If User B 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 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 proxy server must forward 200 OK responses upstream.</p>
15	200 OK—PS -> GW1	The proxy server forwards the 200 OK response that it received from GW2 to GW1.
16	Connect—GW1 -> PBX A	GW1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
17	Connect ACK—PBX A -> GW1	PBX A acknowledges GW1’s Connect message.

Successful Call Flow Scenarios

Step	Action	Description
18	ACK—GW1 -> PS	GW1 sends a SIP ACK request to the proxy server. The ACK request confirms that GW1 has received the 200 OK response from the proxy server. The ACK request might contain a message body with the final session description to be used by User B. If the message body of the ACK request is empty, User B uses the session description in the INVITE request.
19	ACK—PS -> GW2	Depending on the values in the To, From, CSeq, and Call-ID field, the proxy server might process the ACK request locally or proxy it. If the fields in the ACK request match those in previous requests processed by the proxy server, the server proxies the ACK request. If there is no match, the ACK request is proxied as if it were an INVITE request. The proxy server forwards GW1's ACK response to GW2.
20	Connect ACK—GW2 -> PBX B	GW2 acknowledges PBX B's Connect message. The call session is now active. Note The 2-way voice path is established directly between GW1 and GW2; not via the proxy server.
21	Disconnect—PBX B -> GW2	After the call is completed, PBX B sends a Disconnect message to GW2. The Disconnect message starts the call session termination process.
22	BYE—GW2 -> PS	GW2 sends a SIP BYE request to the 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 User B's SIP URL.
23	BYE—PS -> GW1	The proxy server forwards the SIP BYE request to GW1.
24	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
25	Release—GW2 -> PBX B	After the call is completed, GW2 sends a Release message to PBX B.
26	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
27	200 OK—GW1 -> PS	GW1 sends a SIP 200 OK response to the proxy server. The 200 OK response notifies GW2 that GW1 has received the BYE request.
28	200 OK—PS -> GW2	The proxy server forwards the 200 OK response to GW2.
29	Release Complete—PBX B -> GW2	PBX B sends a Release Complete message to GW2.
30	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the call session is terminated.

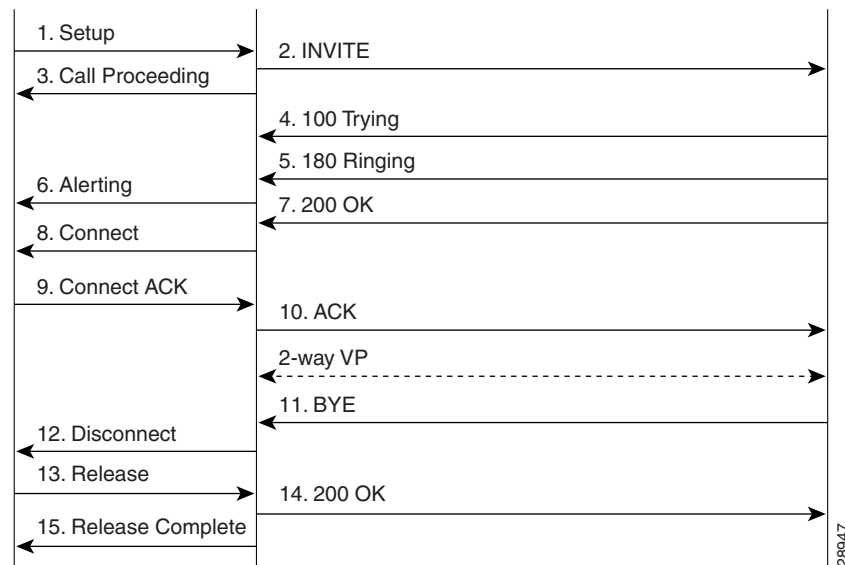
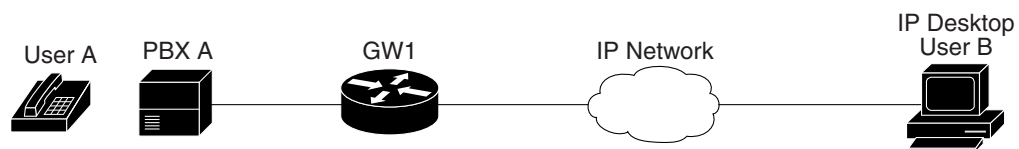
Gateway-to-IP Desktop Call

Figure 4 illustrates a successful gateway-to-IP desktop PC 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 GW1 (SIP Gateway) via a T1/E1. GW1 is using a proxy server. User B is located at an IP desktop. GW1 is connected to the IP desktop over an IP network.

The call flow is as follows:

- 1 User A calls User B's desktop PC.
- 2 User B answers the call.
- 3 User B hangs up.

Figure 4 Gateway-to-IP Desktop Call—Successful Setup and Disconnect



Successful Call Flow Scenarios

Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between the PBX and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> Desktop	<p>GW1 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. GW1 sends a SIP INVITE request to the address it receives as the dial peer which, in this scenario, is the IP desktop.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The IP address of the desktop 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 GW is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	100 Trying—Desktop -> GW1	The IP desktop sends a SIP 100 Trying response to GW1. The 100 Trying response indicates that the INVITE request has been received by the IP desktop.
5	180 Ringing—Desktop -> GW1	The IP desktop sends a SIP 180 Ringing response to GW1. The 180 Ringing response indicates that the user is being alerted.
6	Alerting—GW1 -> PBX A	GW1 sends an Alert message to User A. The Alert message indicates that GW1 has received a 180 Ringing response from the IP desktop. User A hears the ringback tone that indicates that User B is being alerted.
7	200 OK—Desktop -> GW1	The IP desktop sends a SIP 200 OK response to GW1. The 200 OK response notifies GW1 that the connection has been made.
8	Connect—GW1 -> PBX A	GW1 sends a Connect message to PBX A. The Connect message notifies PBX A that the connection has been made.
9	Connect ACK—PBX A -> GW1	PBX A acknowledges GW1's Connect message.
10	ACK—GW1 -> Desktop	GW1 sends a SIP ACK request to the IP desktop. The ACK request confirms that User A has received the 200 OK response. The call session is now active.
11	BYE—Desktop -> GW1	User B terminates the call session at his IP desktop and the IP desktop sends a SIP BYE request to GW1. The BYE request indicates that User B wants to release the call.
12	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
13	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
14	200 OK—GW1 -> Desktop	GW1 sends a SIP 200 OK response to the IP desktop. The 200 OK response notifies the desktop that GW1 has received the BYE request.
15	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the call session is terminated.

Call Flow Scenarios for Failed Calls

This section includes call flows for scenarios in which the call setup has failed because of one of the following reasons:

- The called user is busy—A SIP 486 Busy Here response is received.
- The called user does not answer and the request times out—A SIP CANCEL request is sent.
- A client, server, or global error occurs—A SIP 4xx, 5xx, or 6xx failure response is received.

The network configurations in which the call flow scenarios occur are the following:

- Gateway-to-gateway calls
- Gateway-to-gateway calls via a redirect server
- Gateway-to-gateway calls via proxy server
- Gateway-to-IP desktop calls

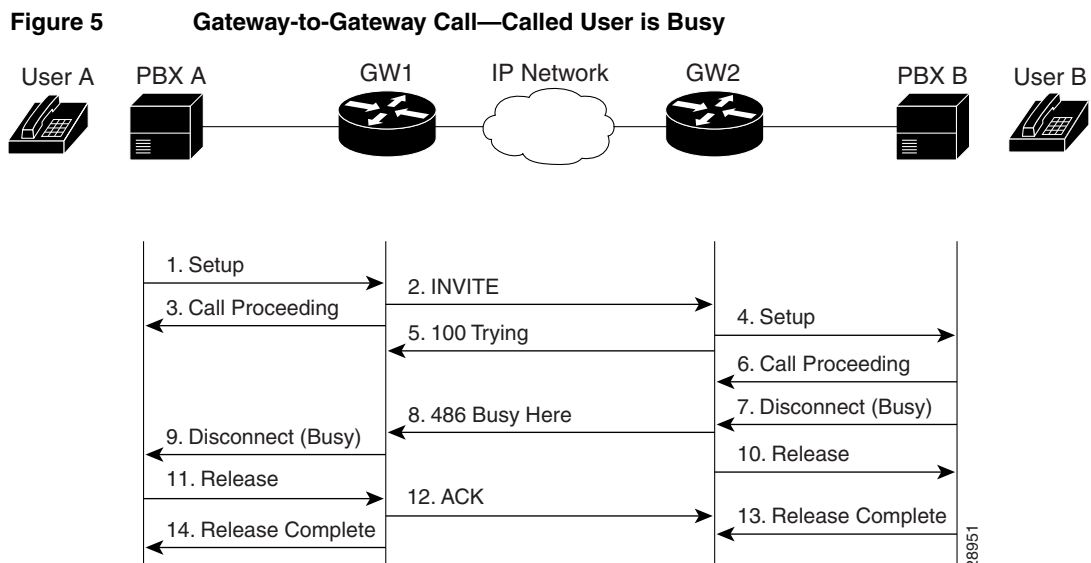
Gateway-to-Gateway Calls

This section describes the call flows for failed gateway-to-gateway calls. In the following call flows, the network configuration is the same as the network configuration outlined in the “Gateway-to-Gateway Call” section on page 2. However, instead of successfully establishing a call session, one of the following situations occurs:

- The Called User is Busy, page 13
- The Called User Does Not Answer, page 15
- A Client, Server, or Global Error Has Occurred, page 16

The Called User is Busy

Figure 5 illustrates the call flow in which User A initiates a call to User B and receives a SIP 486 Busy Here response.



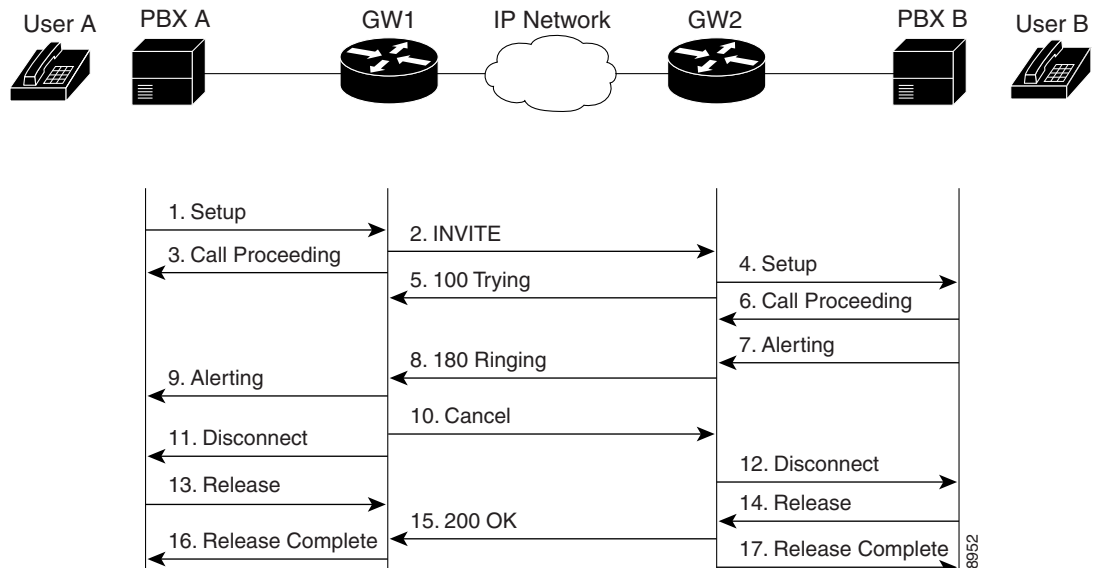
Call Flow Scenarios for Failed Calls

Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> GW2	<p>GW1 sends a SIP INVITE request to GW2. 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 GW1 is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	Setup—GW2 -> PBX B	GW2 receives the INVITE request from GW1 and initiates a Call Setup with User B via PBX B.
5	100 Trying—GW2 -> GW1	GW2 sends a SIP 100 Trying response to the INVITE request sent by GW1. The 100 Trying message indicates that the INVITE request has been received by GW2 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 -> GW2	PBX B sends a Call Proceeding message to GW2 to acknowledge the Call Setup request.
7	Disconnect (Busy)—PBX B -> GW2	PBX B sends a Disconnect message to GW2. 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—GW2 -> GW1	GW2 maps the Release message cause code (Busy) to the SIP 486 Busy response and sends the response to GW1. 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)—GW1 -> PBX A	GW1 sends a Release message to PBX A. User A hears a busy tone.
10	Release—GW2 -> PBX B	GW2 sends a Release message to PBX B.
11	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
12	ACK—GW1 -> GW2	GW1 sends a SIP ACK request to GW2. The ACK request confirms that the 200 OK response has been received.
13	Release Complete—PBX B -> GW2	PBX B sends a Release Complete message to GW2.
14	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the call session attempt is terminated.

The Called User Does Not Answer

Figure 6 illustrates the call flow in which User A initiates a call to User B and the request times out and is ended with a SIP CANCEL request.

Figure 6 Gateway-to-Gateway Call—Called User Does Not Answer



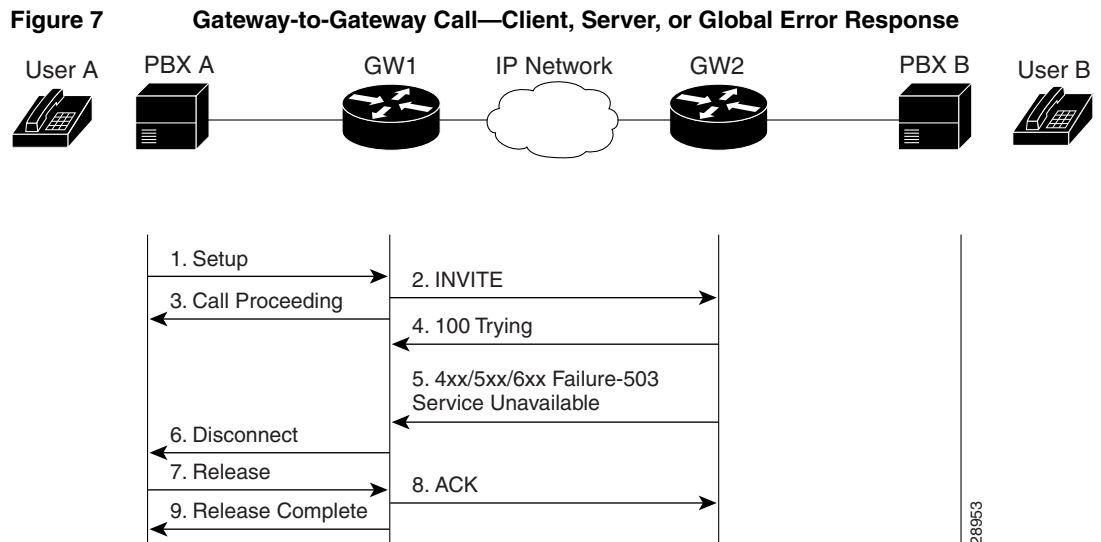
Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> GW2	GW1 sends a SIP INVITE request to GW2. The INVITE request is an invitation to User B to 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. 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 GW1 is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	Setup—GW2 -> PBX B	GW2 receives the INVITE request from GW1 and initiates a Call Setup with User B via PBX B.
5	100 Trying—GW2 -> GW1	GW2 sends a SIP 100 Trying response to the INVITE request sent by GW1. The 100 Trying response indicates that the INVITE request has been received by GW2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.

Call Flow Scenarios for Failed Calls

Step	Action	Description
6	Call Proceeding—PBX B -> GW2	PBX B sends a Call Proceeding message to GW2 to acknowledge the Call Setup request.
7	Alerting—PBX B -> GW2	PBX B sends an Alert message to GW2. User B's phone begins to ring.
8	180 Ringing—GW2 -> GW1	GW2 sends a SIP 180 Ringing response to GW1. The 180 Ringing response indicates that GW2 has located, and is trying to alert, User B.
9	Alerting—GW1 -> PBX A	GW1 sends an Alert message to PBX A. User A hears a ringback tone.
10	Cancel (ring timeout)—GW1 -> GW2	Because GW2 did not return an appropriate response within the time allocated in the INVITE request, GW1 sends a SIP CANCEL request to GW2. A CANCEL request cancels a pending request with the same Call-ID, To, From, and CSeq header field values.
11	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
12	Disconnect—GW2 -> PBX B	GW2 sends a Disconnect message to PBX B.
13	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
14	Release—PBX B -> GW2	PBX B sends a Release message to GW2.
15	200 OK—GW2 -> GW1	GW2 sends a SIP 200 OK response to GW1. The 200 OK response confirms that the Cancel request has been received.
16	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A.
17	Release Complete—GW2 -> PBX B	GW2 sends a Release Complete message to PBX B and the call session attempt is terminated.

A Client, Server, or Global Error Has Occurred

Figure 7 illustrates the call flow in which User A initiates a call to User B and receives a class 4xx, 5xx, or 6xx response. In the following scenario, there are no more channels available on GW2. Therefore, GW2 refuses the connection and sends a SIP 503 Service Unavailable response.



Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> GW2	<p>GW1 sends a SIP INVITE request to GW2. 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 GW1 is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	100 Trying—GW2 -> GW1	GW2 sends a SIP 100 Trying response to the INVITE request sent by GW1. The 100 Trying message indicates that the INVITE request has been received by GW2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
5	Class 4xx/5xx/6xx Failure—GW2 -> GW1	<p>GW2 determines that it does not have any more channels available, refuses the connection, and sends a SIP 503 Service Unavailable response to GW1.</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> <ul style="list-style-type: none"> • If GW2 sends a class 4xx failure response (a definite failure response that is a client error), the request will not be retried without modification. • If GW2 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. • If GW2 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>Note The call failure on GW2 might occur before a proceeding indication from PBX B. In that case a SIP failure response is sent before the SIP 100 Trying response.</p>
6	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
7	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
8	ACK—GW1 -> GW2	GW1 sends a SIP ACK request to GW2. The ACK request confirms that the 200 OK response has been received.
9	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the call session attempt is terminated.

Gateway-to-Gateway Calls via a Redirect Server

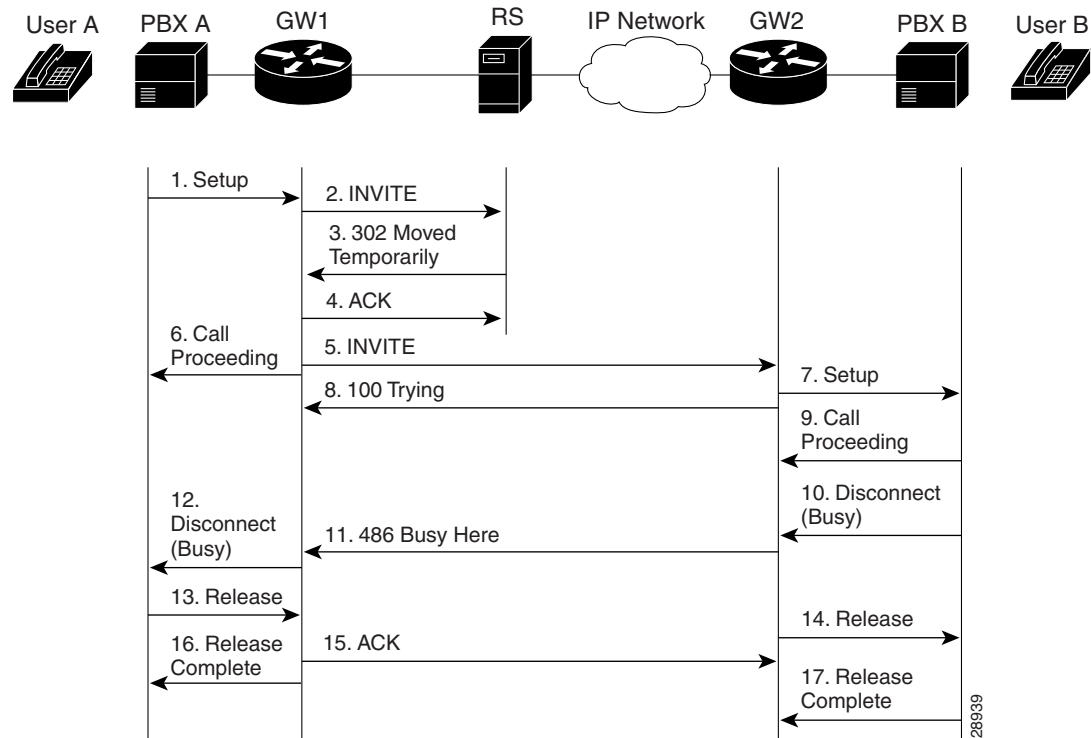
This section describes the call flows for gateway-to-gateway calls via a redirect server that have failed. In the following call flows, the network configuration is the same as the network configuration outlined in the “Gateway-to-Gateway Call via Redirect Server” section on page 5. However, instead of successfully establishing a call session, one of the following situations occurs:

- The Called User is Busy, page 18
- The Called User Does Not Answer, page 20
- A Client, Server, or Global Error Has Occurred, page 22

The Called User is Busy

Figure 8 illustrates the call flow in which User A initiates a call to User B and receives a SIP 486 Busy Here response.

Figure 8 Gateway-to-Gateway Call via a Redirect Server—Called User is Busy



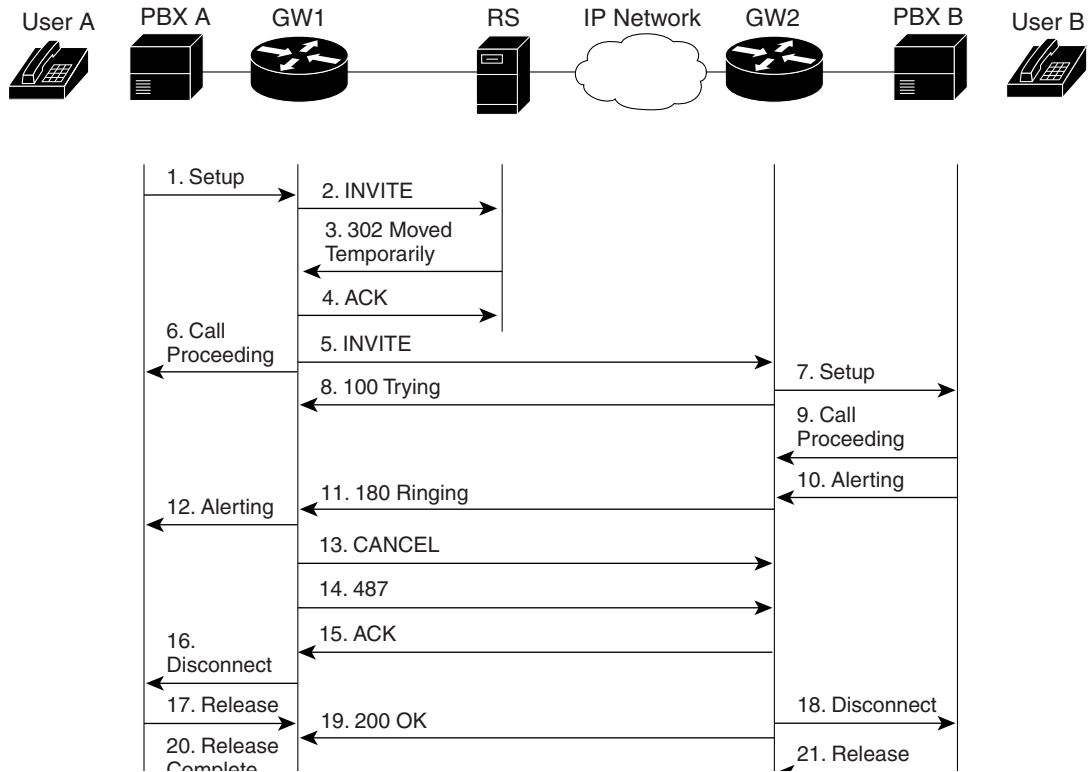
Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—GW1 -> RS	<p>GW1 sends a SIP INVITE request to the 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 GW1 is prepared to receive the RTP data is specified.
3	300 Multiple Choice—RS -> GW1	The redirect server sends GW1 a SIP 300 Multiple Choice response. The 300 Multiple Choice response indicates that the 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 redirect server returns these possible addresses to GW1 in the 300 Multiple Choice response.
4	ACK—GW1 -> RS	GW1 acknowledges the 300 Multiple Choice response with a SIP ACK request.
5	INVITE—GW1 -> GW2	GW1 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—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
7	Setup—GW2 -> PBX B	GW2 receives the INVITE request from GW1 and initiates a Call Setup with User B via PBX B.
8	100 Trying—GW2 -> GW1	GW2 sends a SIP 100 Trying response to the INVITE request sent by GW1. The 100 Trying response indicates that the INVITE request has been received by GW2 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 -> GW2	PBX B sends a Call Proceeding message to GW2 to acknowledge the Call Setup request.
10	Disconnect (Busy)—PBX B -> GW2	PBX B sends a Disconnect message to GW2. 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—GW2 -> GW1	GW2 maps the Release message cause code (Busy) to the SIP 486 Busy response and sends the response to GW1. 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)—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A. User A hears a busy tone.
13	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
14	Release—GW2 -> PBX B	GW1 sends a Release message to PBX B.
15	ACK—GW1 -> GW2	GW1 sends a SIP ACK request to GW2. The ACK request confirms that the 486 Busy Here response has been received.
16	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the call session attempt is terminated.
17	Release Complete—PBX B -> GW2	PBX B sends a Release Complete message to GW2.

The Called User Does Not Answer

Figure 9 illustrates the call flow in which User A initiates a call to User B and the request times out and is ended with a SIP CANCEL request.

Figure 9 Gateway-to-Gateway Call via a Redirect Server—Called User is Does Not Answer



Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> RS	GW1 sends a SIP INVITE request to the 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. 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 GW1 is prepared to receive the RTP data is specified.

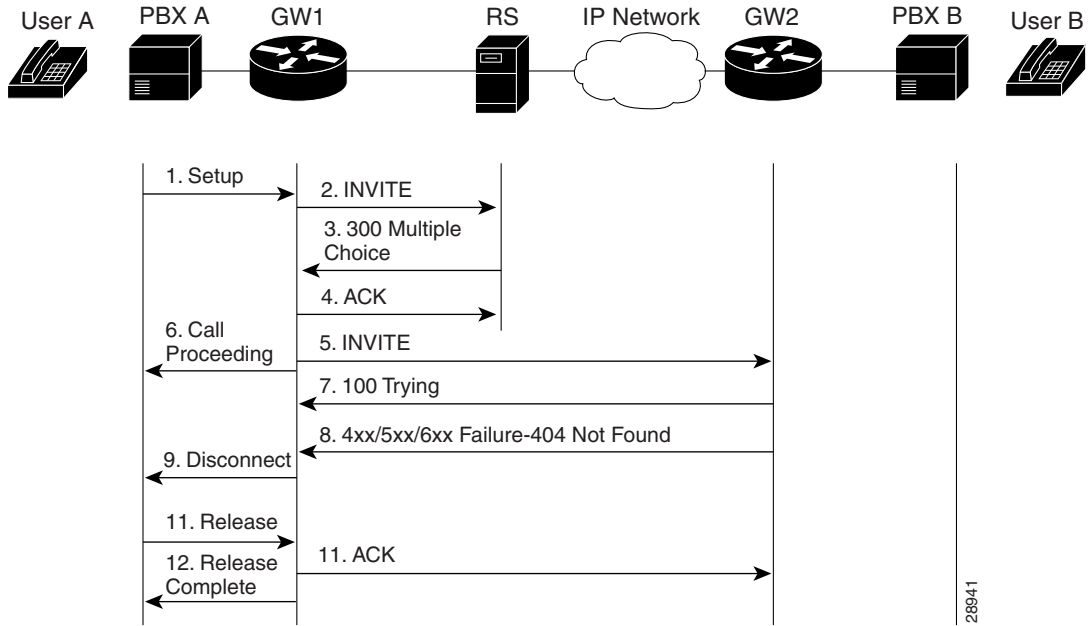
Step	Action	Description
3	300 Multiple Choice—RS -> GW1	The redirect server sends GW1 a SIP 300 Multiple Choice response. The 300 Multiple Choice response indicates that the redirect server accepted the INVITE request, contacted a location server with all or part of User B's SIP URL, and the location server provided a list of alternative locations where User B might be located. The redirect server returns these possible addresses to User A in the 300 Multiple Choice response.
4	ACK—GW1 -> RS	GW1 acknowledges the 300 Multiple Choice response with a SIP ACK request.
5	INVITE—GW1 -> GW2	GW1 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—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
7	Setup—GW2 -> PBX B	GW2 receives the INVITE request from GW1 and initiates a Call Setup with User B via PBX B.
8	100 Trying—GW2 -> GW1	GW2 sends a SIP 100 Trying response to the INVITE request sent by GW1. The 100 Trying message indicates that the INVITE request has been received by GW2 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 -> GW2	PBX B sends a Call Proceeding message to GW2 to acknowledge the Call Setup request.
10	Alerting—PBX B -> GW2	PBX B sends an Alert message to GW2. User B's phone begins to ring.
11	180 Ringing—GW2 -> GW1	GW2 sends a SIP 180 Ringing response to GW1. The 180 Ringing response indicates that GW2 has located, and is trying to alert, User B.
12	Alerting—GW1 to PBX A	GW1 sends an Alert message to PBX A.
13	CANCEL (Ring Timeout)—GW1 -> GW2	Because GW2 did not return an appropriate response within the time allocated in the INVITE request, GW1 sends a SIP CANCEL request to GW2. A CANCEL request cancels a pending request with the same Call-ID, To, From, and CSeq header field values.
14	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
15	Release—PBX A to GW1	PBX A sends a Release message to GW1.
16	Disconnect—GW2 -> PBX B	GW2 sends a Disconnect message to PBX B.
17	200 OK—GW1 -> GW2	GW1 sends a SIP 200 OK response to GW2. The 200 OK response confirms that the CANCEL request has been received.
18	Release Complete—PBX A -> GW1	PBX A sends a Release Complete message to GW1 and the call session attempt is terminated.
19	Release—PBX B -> GW2	PBX B sends a Release message to GW2.
20	Release Complete—GW2 -> PBX B	GW2 sends a Release Complete message to PBX B.

A Client, Server, or Global Error Has Occurred

Figure 10 illustrates the call flow in which User A initiates a call to User B and receives a class 4xx, 5xx, or 6xx response.

In this scenario, GW2 determines that User B does not exist at the domain specified in the INVITE request sent by GW1. GW2 refuses the connection and sends GW1 a SIP 404 Not Found response.

Figure 10 Gateway-to-Gateway Call via a Redirect Server—Client, Server, or Global Error Response



Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> RS	<p>GW1 sends a SIP INVITE request to the 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’s 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 GW1 is prepared to receive the RTP data is specified.

Step	Action	Description
3	300 Multiple Choice—RS -> GW1	The redirect server sends GW1 a SIP 300 Multiple Choice response. The 300 Multiple Choice response indicates that the 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 redirect server returns these possible addresses to User A in the 300 Multiple Choice response.
4	ACK—GW1 -> RS	GW1 acknowledges the 300 Multiple Choice response with a SIP ACK request.
5	INVITE—GW1 -> GW2	GW1 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—GW1 -> GW2	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
7	100 Trying—GW2 -> GW1	GW2 sends a SIP 100 Trying response to the INVITE request sent by GW1. The 100 Trying message indicates that the INVITE request has been received by GW2 but that User B has not yet been located and that some unspecified action, such as a database consultation, is taking place.
8	Class 4xx/5xx/6xx Failure—GW2 -> GW1	<p>GW2 determines that User B does not exist at the domain specified in the INVITE request sent by GW1. GW2 refuses the connection and sends a SIP 404 Not Found response to GW1. The 404 Not Found response is a class 4xx failure response. Depending on which class the failure response is, the call actions differ.</p> <ul style="list-style-type: none"> • If GW2 sends a class 4xx failure response (a definite failure response that is a client error), the request will not be retried without modification. • If GW2 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. • If GW2 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.
9	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
10	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
11	ACK—GW1 -> GW2	GW1 sends a SIP ACK request to GW2. The ACK request confirms that the failure response has been received.
12	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the call session attempt is terminated.

Gateway-to-Gateway Calls via a Proxy Server

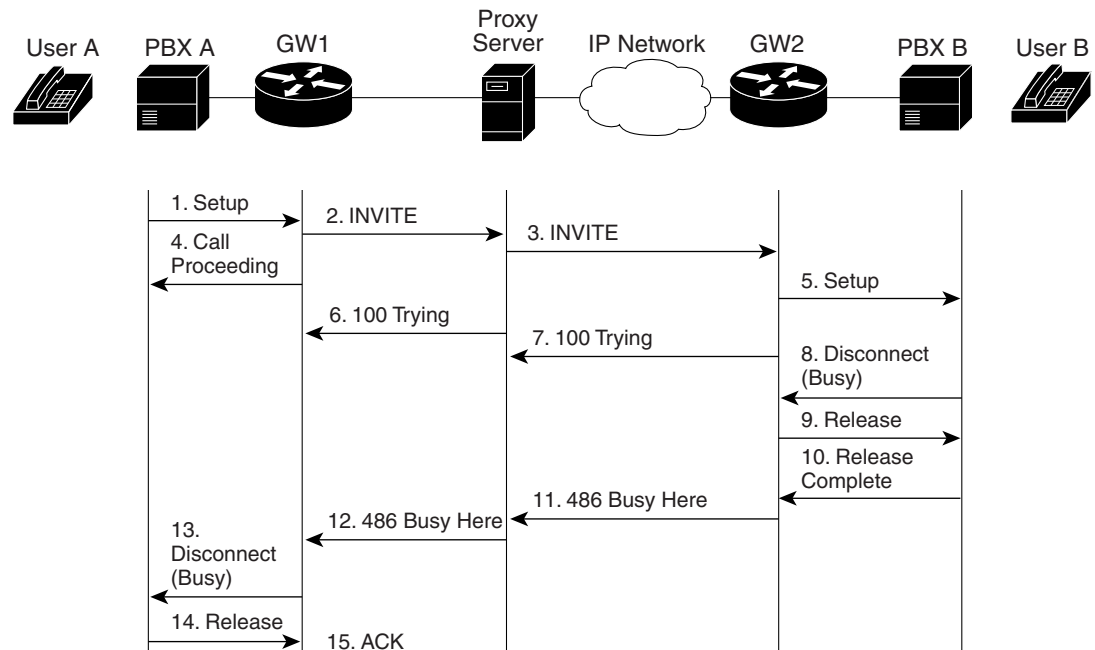
This section describes the call flows for gateway-to-gateway calls via a proxy server that have failed. In the following call flows, the network configuration is the same as the network configuration outlined in the “Gateway-to-Gateway Call via a Proxy Server” section on page 7. However, instead of successfully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

- The Called User is Busy, page 24
- A Client or Server Error Has Occurred, page 26
- A Global Error Has Occurred, page 28

The Called User is Busy

Figure 11 illustrates the call flow in which User A initiates a call to User B and receives a SIP 486 Busy Here response.

Figure 11 Gateway-to-Gateway Call via a Proxy Server—Called User is Busy



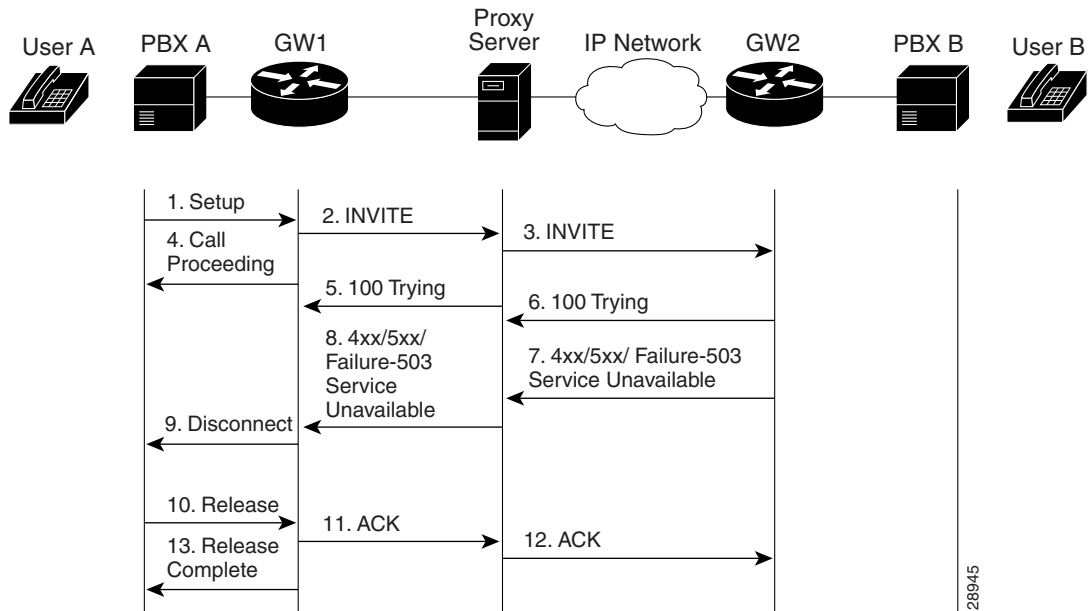
Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> PS	<p>GW1 sends a SIP INVITE request to the proxy server. The INVITE request is an invitation 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 GW1 is prepared to receive the RTP data is specified.
3	INVITE—PS -> GW2	The proxy server checks whether its 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 GW1, 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 GW2.
4	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
5	Setup—GW2 -> PBX B	GW2 receives the INVITE request from the proxy server and initiates a Call Setup with User B via PBX B.
6	100 Trying—PS -> GW1	The proxy server sends a SIP 100 Trying response to GW1.
7	100 Trying—GW2 -> PS	GW2 sends a SIP 100 Trying response to the proxy server.
8	Release Complete (Busy)—PBX B -> GW2	PBX B sends a Release Complete message to GW2. 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—GW2 -> PS	<p>GW2 maps the Release message cause code (Busy) to the SIP 486 Busy response and sends the response to the 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.</p> <p>The proxy server must send a SIP ACK request upon receiving a class 4xx failure response.</p>
10	486 Busy Here—PS -> GW1	The proxy server forwards the SIP 486 Busy response to GW1.
11	Disconnect (Busy)—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
12	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
13	ACK—GW1 -> PS	GW1 sends an SIP ACK request to the proxy server.
14	ACK—PS -> GW2	The proxy server forwards the SIP ACK request to GW2. The ACK request confirms that the 486 Busy Here response has been received.
15	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the call session attempt is terminated.

A Client or Server Error Has Occurred

Figure 12 illustrates the call flow in which User A initiates a call to User B and receives a class 4xx or 5xx response from User B (via GW2).

In the following scenario, there are no more channels available on GW2. Therefore, GW2 refuses the connection and sends a SIP 503 Service Unavailable response.

Figure 12 Gateway-to-Gateway Call via a Proxy Server—Client or Server Error Response



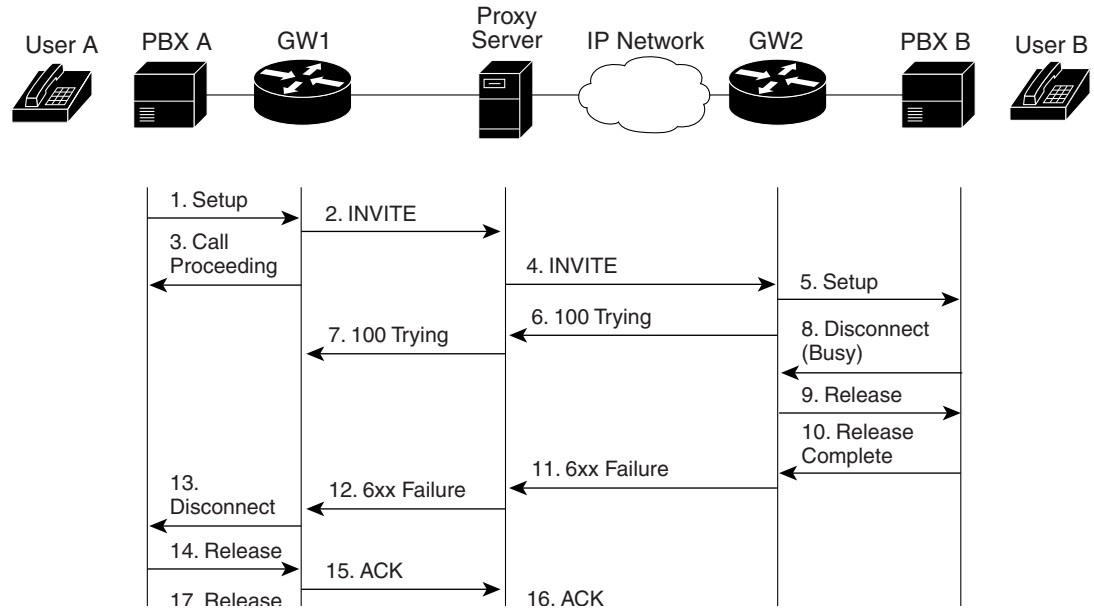
Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> PS	GW1 sends a SIP INVITE request to the proxy server. The INVITE request is an invitation to 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. 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. • 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 GW1 is prepared to receive the RTP data is specified.
3	INVITE—PS -> GW2	The 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 GW1, 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 GW2.
4	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.

Step	Action	Description
5	100 Trying—PS -> GW1	The proxy server sends a SIP 100 Trying response to GW1.
6	100 Trying—GW2 -> PS	GW2 sends a SIP 100 Trying response to the proxy server.
7	Class 4xx/5xx/6xx Failure—GW2 -> PS	<p>GW2 determines that it does not have any more channels available, refuses the connection, and sends a SIP 503 Service Unavailable response to GW1.</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> <ul style="list-style-type: none"> • If GW2 sends a class 4xx or 5xx failure response to the proxy server. Depending on which class the failure response is, the call actions differ. • If GW2 sends a class 4xx failure response (a definite failure response that is a client error), the request will not be retried without modification and the proxy server must send an ACK request. • If GW2 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 and again, the proxy server must send an ACK request.
8	Class 4xx/5xx/6xx Failure—PS -> GW1	The proxy server forwards the SIP 503 Service Unavailable response to GW1.
9	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
10	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
11	ACK—GW1 -> PS	GW1 sends a SIP ACK request to the proxy server.
12	ACK—PS -> GW2	The proxy server forwards the SIP ACK request to GW2. The ACK request confirms that the 486 Busy Here response has been received.
13	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the call session attempt is terminated.

A Global Error Has Occurred

Figure 13 illustrates the call flow in which User A initiates a call to User B and receives a class 6xx response.

Figure 13 Gateway-to-Gateway Call via a Proxy Server—Global Error Response



Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> PS	GW1 sends a SIP INVITE request to the 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. 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 GW1 is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	INVITE—PS -> GW2	The proxy server checks whether its 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 GW1, 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 GW2.
5	Setup—GW2 -> PBX B	GW2 receives the INVITE request from the proxy server and initiates a Call Setup with User B via PBX B.

Step	Action	Description
6	100 Trying—GW2 -> PS	GW2 sends a SIP 100 Trying response to the proxy server. The proxy server might or might not forward the 100 Trying response to GW1.
7	100 Trying—PS -> GW1	The proxy server forwards the SIP 100 Trying response to GW1.
8	Release Complete—PBX B -> GW2	PBX B sends a Release Complete message to GW2. The Release Complete message starts the call session termination process.
9	6xx Failure—GW2 -> GW1	GW2 sends a class 6xx failure response (a global error) to GW1. 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 proxy server must forward all class 6xx failure responses to the client and send an ACK.
10	6xx Failure—PS -> GW1	The proxy server forwards the 6xx failure to GW1.
11	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to PBX A.
12	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
13	ACK—GW1 -> PS	GW1 sends a SIP ACK request to the proxy server.
14	ACK—PS -> GW2	The proxy server sends a SIP ACK request to GW2. The ACK request confirms that the 486 Busy Here response has been received.
15	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to PBX A and the call session attempt is terminated.

Gateway-to-IP Desktop

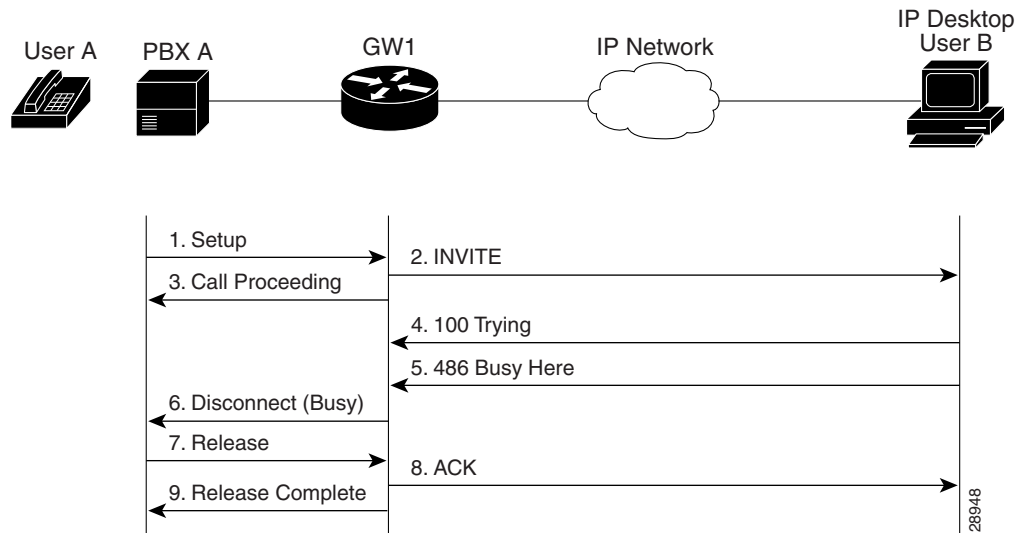
This section describes the call flows for failed gateway-to-IP desktop calls. In the following call flows, the network configuration is the same as the network configuration outlined in the “Gateway-to-IP Desktop Call” section on page 11. However, instead of successfully establishing a call session, one of the following situations occurs:

- The Called User is Busy, page 30
- The Called User Does Not Answer, page 32
- A Client, Server, or Global Error Has Occurred, page 34

The Called User is Busy

Figure 14 illustrates the call flow in which User A initiates a call to User B and receives a SIP 486 Busy Here response.

Figure 14 Gateway-to-IP Desktop Call—Called User is Busy



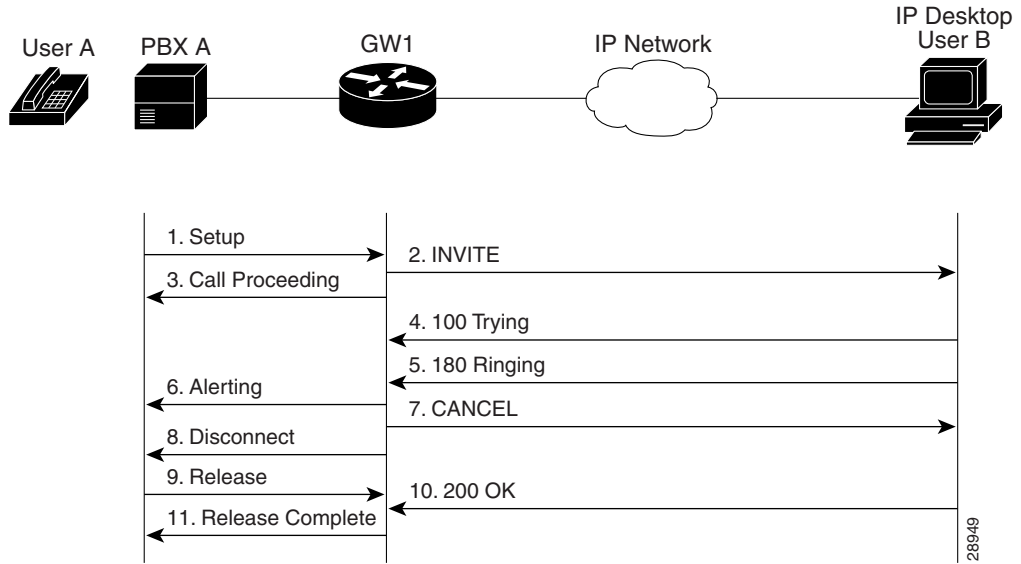
Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between the PBX A and the GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.

Step	Action	Description
2	INVITE—GW1 -> Desktop	<p>GW1 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. The GW1 sends a SIP INVITE request to the address it receives as the dial peer which, in this scenario, is the IP desktop.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The IP address of the desktop 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 GW is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	100 Trying—Desktop -> GW1	The IP desktop sends a SIP 100 Trying response to the GW1. The 100 Trying response indicates that the INVITE request has been received by the IP desktop.
5	486 Busy Here—Desktop -> GW1	The IP desktop sends a SIP 480 Busy Here response to the GW1. 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)—GW1 -> PBX A	The GW1 sends a Disconnect message to the PBX A.
7	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
8	ACK—GW1 -> Desktop	The GW1 sends a SIP ACK request to the IP desktop. The ACK request confirms that User A has received the 486 Busy Here response. The call session attempt is now being terminated.
9	Release Complete—GW1 -> PBX A	The GW1 sends a Release Complete message to the PBX A and the call session attempt is terminated.

The Called User Does Not Answer

Figure 15 illustrates the call flow in which User A initiates a call to User B and the request times out and is ended with a SIP CANCEL request.

Figure 15 Gateway-to-IP Desktop Call—Called User Does Not Answer

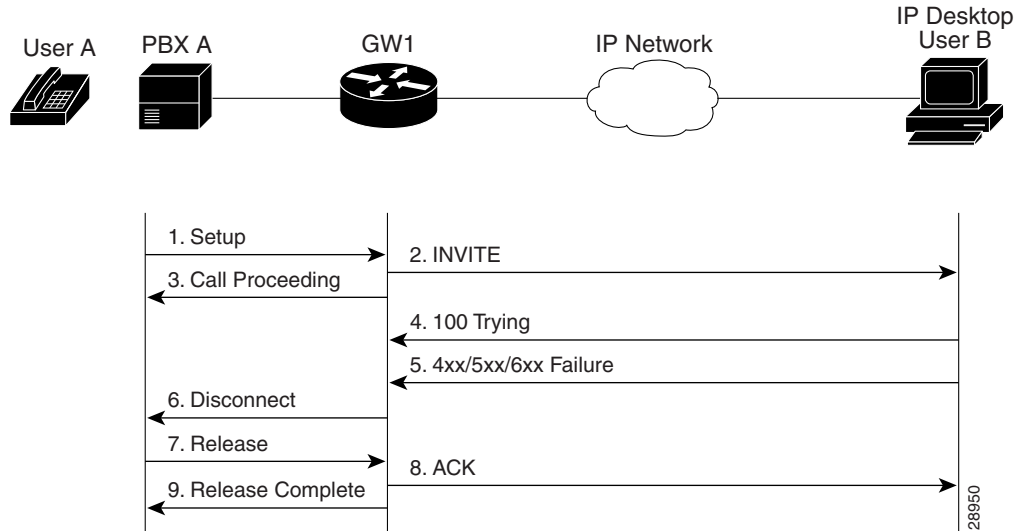


Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between the PBX A and the GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> Desktop	<p>GW1 maps the SIP URL phone number to a dial-peer. The dial-peer includes the IP address and the port number of the SIP enabled entity to contact. The GW1 sends a SIP INVITE request to the address it receives as the dial peer which, in this scenario, is the IP desktop.</p> <p>In the INVITE request:</p> <ul style="list-style-type: none"> • The IP address of the desktop 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 GW is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	100 Trying—Desktop -> GW1	The IP desktop sends a SIP 100 Trying response to the GW1. The 100 Trying response indicates that the INVITE request has been received by the IP desktop.
5	180 Ringing—Desktop -> GW1	The IP desktop sends a SIP 180 Ringing response to the GW1. The 180 Ringing response indicates that the user is being alerted.
6	Alerting—GW1 -> PBX A	GW1 sends an Alert message to PBX A.
7	CANCEL (Ring Timeout)—GW1 -> Desktop	Because GW1 did not return an appropriate response within the time allocated in the INVITE request, GW1 sends a SIP CANCEL request to GW2. A CANCEL request cancels a pending request with the same Call-ID, To, From, and CSeq header field values.
8	Disconnect—GW1 -> PBX A	GW1 sends a Disconnect message to the PBX A.
9	Release Complete—GW1 -> PBX A	GW1 sends a Release Complete message to the PBX A and the call session attempt is terminated.
10	200 OK—GW1 -> Desktop	The GW1 sends a SIP 200 OK response to the IP desktop. The 200 OK response confirms that User A has received the 486 Busy Here response. The call session attempt is now being terminated.

A Client, Server, or Global Error Has Occurred

Figure 16 illustrates the call flow in which User A initiates a call to User B and receives a class 4xx, 5xx, or 6xx response.

Figure 16 Gateway-to-IP Desktop Call—Client, Server, or Global Error Response



Step	Action	Description
1	Setup—PBX A -> GW1	Call Setup is initiated between the PBX A and GW1. The Call Setup includes the standard transactions that take place as User A attempts to call User B.
2	INVITE—GW1 -> Desktop	GW1 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. GW1 sends a SIP INVITE request to the address it receives as the dial peer which, in this scenario, is the IP desktop. In the INVITE request: <ul style="list-style-type: none"> • The IP address of the desktop 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 GW is prepared to receive the RTP data is specified.
3	Call Proceeding—GW1 -> PBX A	GW1 sends a Call Proceeding message to PBX A to acknowledge the Call Setup request.
4	100 Trying—Desktop -> GW1	The IP desktop sends a SIP 100 Trying response to the GW1. The 100 Trying response indicates that the INVITE request has been received by the IP desktop.

Step	Action	Description
5	4xx/5xx/6xx Failure—Desktop -> GW1	<p>The IP desktop sends a class 4xx, 5xx, or class 6xx failure response to the GW1. Depending on which class the failure response is, the call actions differ.</p> <ul style="list-style-type: none"> • If the IP desktop sends a class 4xx failure response (a definite failure response that is a client error), the request will not be retried without modification. • If the IP desktop 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. • If the IP desktop 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.
6	Disconnect—GW1 -> PBX A	The GW1 sends a Release message to the PBX A.
7	Release—PBX A -> GW1	PBX A sends a Release message to GW1.
8	ACK—GW1 -> Desktop	The GW1 sends a SIP ACK request to the IP desktop. The ACK request confirms that User A has received the 486 Busy Here response. The call session attempt is now being terminated.
9	Release Complete—GW1 -> PBX A	The GW1 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 36
- SIP Methods, page 36
- SIP Responses, page 36
- SIP Header Fields, page 40
- SIP Transport Layer Protocols, page 41
- SIP Security, page 41
- SIP Session Description Protocol (SDP) Usage, page 41
- SIP DNS Records Usage, page 42

SIP Functions

Function	Supported?
User Agent Client (UAC)	Yes
User Agent Server (UAS)	Yes
Proxy Server	The SIP gateway does not have the proxy or redirect server functionality, but can work with an external third-party proxy or redirect server.
Redirect Server	

SIP Methods

There are six methods used by the SIP gateway.

Method	Supported?	Comments
INVITE	Yes	The INVITE support handles an initial INVITE for the same Call ID. The INVITE permits CODEC changes.
ACK	Yes	
OPTIONS	Yes	The SIP gateway does not generate OPTIONS, however, it does responds to OPTIONS methods.
BYE	Yes	
CANCEL	Yes	
REGISTER	NA	In SIP, there is no requirement for a SIP gateway to register using this method. Therefore, the gateway does not generate or process this method.

SIP Responses

Cisco IOS Release 12.1(1)T supports the following SIP Responses:

- 1xx Response—Information Responses, page 37
- 2xx Response—Successful Responses, page 37
- 3xx Response—Redirection Responses, page 38
- 4xx Response—Request Failure Responses, page 38
- 5xx Response—Server Failure Responses, page 39
- 6xx Response—Global Responses, page 39

1xx Response—Information Responses

1xx Response	Comments
100 Trying	The SIP gateway generates a 100 Trying response for an incoming INVITE. The gateway stops the retransmission of INVITEs once it has received a 100 Trying response. After receiving a 100 Trying response, the gateway waits for a 180 Ringing or a 200 OK response.
180 Ringing	The SIP gateway generates a 180 Ringing response when the called party has been located and is being alerted. On receiving a 180 Ringing response, the gateway waits for a 200 OK response.
181 Call is being forwarded	The SIP gateway does not generate these responses. The gateway processes a 181 Call is being forwarded response the same way that it processes the 100 Trying response.
182 Queued	

2xx Response—Successful Responses

2xx Response	Comments
200 OK	None.

3xx Response—Redirection Responses

3xx Response	Comments
300 Multiple Choices	In 12.1(1), a second contact is tried only if the first contact does not return a 180 Ringing, 200 OK, 486 Busy, or a 600 Busy everywhere response. The SIP gateway does not generate this response. The gateway contacts the new address in the Contact header field.
301 Moved Permanently	
302 Moved Temporarily	
305 Use Proxy	The SIP gateway does not generate these responses. The gateway contacts the new address in the Contact header field.
380 Alternate Service	

4xx Response—Request Failure Responses

4xx Response	Comments
400 Bad Request	The SIP gateway generates a 400 Bad Request response for an erroneous request. For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
401 Unauthorized	The SIP gateway does not generate these 4xx responses. For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
402 Payment Required	
403 Forbidden	
404 Not Found	The SIP gateway generates the 404 Not Found response when it is unable to locate the callee. For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
405 Method Not Allowed	The SIP gateway generates a 405 Method Not Allowed for an invalid method. For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
406 Not Acceptable	The SIP gateway does not generate a 406 Not Acceptable response. For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
407 Proxy Authentication Required	The SIP gateway does not generate these 4xx responses. For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
408 Request Timeout	
409 Conflict	
410 Gone	
411 Length Required	
413 Request Entity Too Large	
414 Request—URL Too Long	
415 Unsupported Media	
420 Bad Extension	For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.

4xx Response	Comments
480 Temporarily Unavailable	The SIP gateway does not generate these 4xx responses. For an incoming response, the gateway initiates a graceful call disconnect (during which the caller hears a busy or fast busy tone) before clearing the call request.
481 Call Leg/Transaction Does Not Exist	
482 Loop Detected	
483 Too Many Hops	
484 Address Incomplete	
485 Ambiguous	
486 Busy Here	

5xx Response—Server Failure Responses

5xx Response	Comments
500 Internal Server Error	The SIP gateway generates the 500, 503, and 505 responses.
501 Not Implemented	
502 Bad Gateway	For an incoming response, the SIP gateway sends a new request if an additional contact address is present. If an additional contact address is not present, the gateway initiates a graceful call disconnect.
503 Service Unavailable	
504 Gateway Timeout	
505 Version Not Supported	

6xx Response—Global Responses

6xx Response	Comments
600 Busy Everywhere	The SIP gateway does not generate these 6xx responses. For an incoming response, the gateway initiates a graceful call disconnect.
603 Decline	
604 Does Not Exist Anywhere	
606 Not Acceptable	

SIP Header Fields

Header Field	Supported?
Accept	Yes
Accept-Encoding	No
Accept-Language	No
Allow	Yes
Authorization	No
Call-ID	Yes
Contact ¹	Yes
Content-Encoding	No
Content-Length	Yes
Content-Type	Yes
Cseq	Yes
Date	Yes
Encryption	No
Expires	No
From	Yes
Hide	No
Max-Forwards	No
Organization	No
Priority	No
Proxy-Authenticate	No
Proxy Authorization	No
Proxy-Require	No
Record-Route	No
Require	Yes
Response-Key	No
Retry-After	No
Route	No
Server	No
Subject	No
Timestamp	No
To	Yes
Unsupported	No
User-Agent	Yes
Via	Yes
Warning	Yes
WWW-Authenticate	No

¹ In Cisco IOS Release 12.1(1), the Contact header is only supported in incoming 3xx responses.

SIP Transport Layer Protocols

Transport Layer Protocol	Supported?	Comments
Unicast UDP	Yes	None.
Multicast UDP	No	There are two applications for Multicast UDP. The first application is multicast registration. Because gateways do not register in the SIP environment, the multicast registration is not needed. The second application is multicast RTP session which is for future use and is not a requirement in Cisco IOS Release 12.1(1)T.
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
a—Session name	No
c—Connection information	Yes
m—Media name and transport address	Yes

SIP DNS Records Usage

DNS Resource Record Type	Supported?
Type A	Yes
Type SRV	Yes

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