SIP Telephony Call Flow Examples

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Abstract

This document gives examples of SIP IP Telephony call flows. Elements in these call flows include SIP User Agents and Clients, SIP Proxy and Redirect Servers, and Gateways to the PSTN (Public Switch Telephone Network). Scenarios include SIP Registration, SIP to SIP calling, SIP to Gateway, Gateway to SIP, and Gateway to Gateway via SIP. Call flow diagrams and message details are shown. PSTN telephony protocols are illustrated using SS7 (Signaling System 7), ISDN (Integrated Services Digital Network) and FGB (Feature Group B) circuit associated signaling. PSTN calls are illustrated using global telephone numbers from the PSTN and from private extensions served on a PBX (Private Branch Exchange).
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Overview

The call flows shown in this document were developed in the design of a carrier-class SIP IP Telephony network. They represent an example minimum set of functionality for SIP to be used in IP Telephony applications.

It is the hope of the authors that this document will be useful for SIP implementors, designers, and protocol researchers alike and will help further the goal of a standard SIP implementation for IP Telephony. It is envisioned that as changes to the standard and additional RFCs are added that this document will reflect those changes and represent the current state of a standard interoperable SIP IP Telephony implementation.

These call flows are based on the current version 2.0 of SIP in RFC2543[2]. Additions and changes to SIP necessary for PSTN interworking are referenced as IETF Internet-Drafts as they are used in the call flows.

Various PSTN signaling protocols are illustrated in this document: SS7 (Signaling System 7), ISDN (Integrated Services Digital Network), and FGB (Feature Group B) circuit associated signaling. They were chosen to illustrate the nature of SIP/PSTN interworking—they are not a complete or even representative set. Also, some details and parameters of these PSTN protocols have been omitted. The intent of this document was not to provide a complete and exact mapping of PSTN protocols to SIP. Rather the emphasis is on the SIP signaling, the message interaction, and the modifications to SIP currently proposed to solve IP Telephony issues.

Finally, these call flows show a minimal implementation. Not even basic telephony features such as call forwarding or call waiting are included. A typical carrier-class implementation of a basic set of telephony features using SIP is described in another document[3].

1.1 General Assumptions

A number of architecture, network, and protocol assumptions underlie the call flows in this document. They are outlined in this section so that they may be taken into consideration. Differences in these assumptions will affect the nature of the call flows.

The authentication of SIP User Agents in these example call flows is performed using SIP Digest.

No authentication of Gateways is shown, since it is assumed that:
. Gateways will only accept calls routed through a trusted Proxy.
. Proxies will perform the Client authentication.
. The Proxy and the Gateway will authenticate each other using IPSec[4].

The SIP Proxy Server has access to a Location Manager and other databases. Information present in the Request-URI and the context (From header) is sufficient to determine to which proxy or gateway the message should be routed. In most cases, a primary and secondary route will be determined in case of Proxy or Gateway failure downstream.

The Proxy Servers in these call flows insert Record-Route headers into requests to ensure that they are in the signaling path for future message exchanges.

Gateways receive enough information in the Request-URI field to determine how to route a call, i.e. what trunk group or link to select, what digits to outpulse, etc.

Gateways provide tones (ringing, busy, etc) and announcements to the PSTN side based on SIP response messages, or pass along audio in-band tones (ringing, busy tone, etc.) in an early media stream to the SIP side.

Two types of Gateways are described in this document:

. Network Gateway. This high port count PSTN gateway originates and terminates calls to the PSTN. It’s use is shared by many customers. Incoming calls from the PSTN have the From header populated with a SIP URL containing the telephone number from the calling party telephone number, if available. A Network Gateway typically uses carrier protocols such as SS7.

. Enterprise Gateway. This low port count PBX (Private Branch Exchange) gateway has trunks or lines for a single customer or user. Incoming calls from the PBX have the From header populated with a provisionable string which uniquely identifies the customer, trunk group, or carrier. This allows private numbers to be interpreted in their correct context. An Enterprise Gateway typically uses SS7, ISDN, circuit associated
signaling, or other PBX interfaces.

The interactions between the Proxy and Gateway can be summarized as follows:

. The SIP Proxy Server performs digit analysis and lookup and locates the correct gateway.
The SIP Proxy Server performs gateway location based on primary and secondary routing.

Digit handling by the Gateways will be as follows:

- Dialed digits received from a Network or Enterprise Gateway will be put in a SIP URL with a telephone number. The number will either be globalized (e.g. sip:+1-314-555-1111@ngw.wcom.com;user=phone) or left as a private number (sip:555-6666,phone-context=p1234@gw.wcom.com;user=phone) which will require interpretation based on From header. The "phone-context=" qualifier is used to interpret the private number. It is used the same as the tag of the same name from the Tel URL draft[5]. However, its use in the user portion of the SIP URL should not require changes to parsers. All Gateways will need to be provisioned to be able to parse the user portion of a Request-URI to determine the customer, trunk group, or circuit referenced.

- The From header will be populated with a SIP URL with a telephone number if it is Calling Party number (CgPN) from the PSTN. If it is an Enterprise Gateway, a provisionable string which uniquely identifies the customer, trunk group, or carrier will be used in the sip URI (e.g. From: sip:ProvisionableString@gw1.wcom.com ;user=phone).

- Note that an alternative to using a SIP URL for telephone numbers is the TEL URL[5]. The major difference between using the SIP URL and the TEL URL is that the SIP URL is routable in a SIP network (resolves down to an IP address) where the TEL URL is not (it just represents digits). For example, a SIP URL can be used in a Contact header, but a TEL URL can not.

These flows show UDP for transport. TCP could also be used.

1.2 Legend for Message Flows

Important Note: In this text version of this Internet Draft, figures containing the message flows have been deleted. They will be present in the next draft of this document. A PostScript (.ps) and Acrobat (.pdf) version are available which contain the figures.

Dashed lines _ represent control messages that are mandatory to the call scenario. These control messages can be SIP or PSTN signaling.

Solid lines _ represent media paths between network elements.

Dashed line with parenthesis around name - represent optional control messages.
Messages are identified in the Figures as F1, F2, etc. This references the message details in the table that follows the Figure. Comments in the message details are shown in the following form:

    /* Comments. */

1.3 SIP Protocol Assumptions

Except for the following, this call flows document uses the April 1999 version 2.0 of SIP defined by RFC2543[2]. The following changes/extensions are assumed throughout:

. A Contact header is included with every INVITE message.

. A Contact header is included in every 200 OK Response.

. The 183 Session Progress response message[5] is used in SIP to Gateway and Gateway to Gateway via SIP calling (Sections 4 and 6). The 183 response with SDP will cause the User Agent to immediately play the SDP media stream to hear in-band call progress information. See Section 4 for more information.

. A Content-Length header is present in every message, set to zero if there is no message body.

. The final entry in a Route header is always the Contact information obtained from the INVITE or 200 OK messages.

. In the SDP message bodies, the time field is "t=0 0". It is expected that an actual SDP message body would have a non-zero start timestamp.
SIP Registration Services

2.1 Success Scenarios

Registration either validates or invalidates a SIP client for user services provided by the SIP proxy and/or SIP server. Additionally, the client provides one or more contact locations to the SIP server with the registration request.

2.1.1 SIP Client New Registration

User B initiates a new SIP session with the SIP server (i.e. the user "logs on to" the SIP server). User B sends a SIP REGISTER request to the SIP server. The request includes the user’s contact list. The SIP server provides a challenge to User B. User B enters her/his valid user ID and password. User B’s SIP client encrypts the user information according to the challenge issued by the SIP server and sends the response to the SIP server. The SIP server validates the user’s credentials. It registers the user in its contact database and returns a response (200 OK) to User B’s SIP client. The response includes the user’s current contact list in Contact headers. The format of the authentication shown is SIP digest as described by RFC2543[2].

Message Details

REGISTER F1
B->SIP server

REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 123456789@here.com
CSeq: 1 REGISTER
Contact: TheLittleGuy <sip:UserB@there.com>
Contact: sip:+1-972-555-2222@gw1.wcom.com;user=phone
Contact: tel:+1-972-555-2222
Content-Length: 0

Unauthorized F2
SIP server-> User B

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 123456789@here.com
CSeq: 1 REGISTER
WWW-Authenticate: Digest realm="MCI WorldCom SIP", domain="wcom.com",
nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="", stale="FALSE",
algorithm="MD5"
Content-Length: 0

REGISTER F3
B->SIP server

REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 123456790@here.com
CSeq: 1 REGISTER
Contact: TheLittleGuy <sip:UserB@there.com>
Contact: sip:+1-972-555-2222@gw1.wcom.com;user=phone
Contact: tel:+1-972-555-2222
Authorization: Digest username="UserB", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="",
url="sip:ss2.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Length: 0

200 OK F4
SIP server
-> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 1234567890@here.com
CSeq: 1 REGISTER
Contact: TheLittleGuy <sip:UserB@there.com>
Contact: sip:+1-972-555-2222@gw1.wcom.com;user=phone
Contact: tel:+1-972-555-2222
Content-Length: 0
2.1.2  
User Cancels Registration

User B wishes to cancel her/his registration with the SIP registrar/redirect server. User B sends a SIP REGISTER request to the SIP server. The request has an expiration period of 0 and applies to all existing contact locations. Since the user already has authenticated with the server, the user supplies authentication credentials with the request and is not challenged by the server. The SIP server validates the user's credentials. It clears the user's contact list, and returns a response (200 OK) to User B's SIP client.

Message Details

REGISTER F1
B--->SIP server

REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 123456791@here.com
CSeq: 1 REGISTER
Expires: 0
Contact: *
Authorization:Digest username="UserB", realm="MCI WorldCom SIP", nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="", url="sip:ss2.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Length: 0

200 OK F2
SIP server
--> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 1234567891@here.com
CSeq: 1 REGISTER
Content-Length: 0
2.1.3  User updates contact list

User B wishes to update the list of addresses where the SIP server will redirect INVITE requests. Note this scenario assumes that Scenario 2.1.1 has taken place, but 2.1.2 has not (i.e. User B currently has 3 contacts registered with SS2.)

User B sends a SIP REGISTER request to the SIP server. User B’s request includes an updated contact list. Since the user already has authenticated with the server, the user supplies authentication credentials with the request and is not challenged by the server. The SIP server validates the user’s credentials. It registers the user in its contact database, updates the user’s contact list, and returns a response (200 OK) to User B’s SIP client. The response includes the user’s current contact list in Contact headers.

Message Details

REGISTER F1
B->SIP server

REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 123456791@here.com
CSeq: 1 REGISTER
Contact: mailto:UserB@there.com
Authorization:Digest username="UserB", realm="MCI WorldCom SIP", nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="", url="sip:ss2.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Length: 0

200 OK F2
SIP server
-> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 1234567891@here.com
CSeq: 1 REGISTER
Contact: TheLittleGuy <sip:UserB@there.com>
Contact: sip:+1-972-555-2222@gw1.wcom.com;user=phone
2.1.4 User Requests Current Contact List

User B sends a register request to the Proxy server containing no Contact headers, indicating the user wishes to query the server for the user’s current contact list. Since the user already has authenticated with the server, the user supplies authentication credentials with the request and is not challenged by the server. The SIP server validates the user’s credentials. It registers the user in its contact database and returns a response (200 OK) to User B’s SIP client. The response includes the user’s current contact list in Contact headers.

Message Details

REGISTER F1
B->SIP server

REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 123456792@here.com
CSeq: 1 REGISTER
Authorization:Digest username="UserB", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1ce4341ae6cbe5a359", opaque="",
uri="sip:ss2.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Length: 0

200 OK F2
SIP server
-> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 1234567892@here.com
CSeq: 1 REGISTER
Contact: TheLittleGuy <sip:UserB@there.com>
Contact: sip:+1-972-555-2222@gw1.wcom.com;user=phone
Contact: tel:+1-972-555-2222
Contact: mailto:UserB@there.com
Content-Length: 0
2.2 Failure Scenarios

2.2.1 Unsuccessful SIP registration

User B sends a SIP REGISTER request to the SIP server. The SIP server provides a challenge to User B. User B enters her/his user ID and password. User B’s SIP client encrypts the user information according to the challenge issued by the SIP server and sends the response to the SIP server. The SIP server attempts to validate the user’s credentials, but they are not valid (the user’s password does not match the password established for the user’s account). The server returns a response (401 Unauthorized) to User B’s SIP client.

Message Details

REGISTER F1
B->SIP server

REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 123456789@here.com
CSeq: 1 REGISTER
Contact: TheLittleGuy <sip:UserB@there.com>
Contact: sip:+1-972-555-2222@gw1.wcom.com;user=phone
Contact: tel:+1-972-555-2222
Content-Length: 0

Unauthorized F2
SIP server-> User B

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 123456789@here.com
CSeq: 1 REGISTER
WWW-Authenticate: Digest realm="MCI WorldCom SIP", domain="wcom.com",
nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="", stale="FALSE",
algorithm="MD5"
Content-Length: 0

REGISTER F3
B->SIP server

REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 123456791@here.com
CSeq: 1 REGISTER
Contact: TheLittleGuy <sip:UserB@there.com>
Contact: sip:+1-972-555-2222@gw1.wcom.com;user=phone
Contact: tel:+1-972-555-2222
Authorization:Digest username="UserB", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="",
uri="sip:ss2.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Length: 0

Note: The response above encodes the incorrect password _IForgot_

Unauthorized F4
SIP server-> User B

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 1234567891@here.com
CSeq: 1 REGISTER
WWW-Authenticate: Digest realm="MCI WorldCom SIP", domain="wcom.com",
nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="", stale="FALSE",
algorithm="MD5"
Content-Length: 0
3 SIP to SIP Dialing

3.1 Success Scenarios

This section details calls between two SIP User Agent Clients (UACs) _User A and User B. User A (TheLittleGuy sip:UserA@here.com) and User B (TheBigGuy sip:UserB@there.com) are assumed to be SIP phones or SIP-enabled devices. Calls route using at least one SIP Proxy server. The successful calls show the initial signaling, the exchange of media information in the form of SDP payloads, the establishment of the media session, then finally the termination of the call.

SIP digest authentication is used by the first Proxy Server to authenticate the caller User A. It is assumed that User B has registered with Proxy Server SS2 as per Section 2.1 to be able to receive the calls.
3.1.1 Successful SIP to SIP through two proxies

In this scenario, User A completes a call to User B using two proxies SS1 and SS2. The initial INVITE (F1) does not contain the Authorization credentials SS1 requires, so a 407 Proxy Authorization response is sent containing the challenge information. A new INVITE (F4) is then sent containing the correct credentials and the call proceeds. The call terminates when User B disconnects by initiating a BYE message.

SS1 inserts a Record-Route header into the INVITE message to ensure that it is present in all subsequent message exchanges. SS2 also inserts itself into the Record-Route header. The ACK (F15) and BYE (F18) both have a Route header.

A tag is inserted by User B in message F9 since the initial INVITE message contains more than one Via header and may have been forked.

Message Details

INVITE F1
A -> Proxy 1

INVITE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
/* Proxy 1 challenges User A for authentication */

407 Proxy Authorization Required F2
Proxy 1 -> User A

SIP/2.0 407 Proxy Authorization Required
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Proxy-Authenticate: Digest realm="MCI WorldCom SIP",
domain="wcom.com", nonce="ea9c8e88df84f1ec4341ae6cbe5a359",
opaque="", stale="FALSE", algorithm="MD5"
Content-Length: 0

ACK F3
A -> Proxy 1

ACK sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* User A responds be re-sending the INVITE with authentication credentials in it. */

INVITE F4
A -> Proxy 1

INVITE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1ec4341ae6cbe5a359", opaque="",
url="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length:132
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Proxy 1 accepts the credentials and forwards the INVITE to Proxy 2. Proxy 1 is assumed to have been authenticated by Proxy 2 using IPSec. Client for A prepares to receive data on port 49170 from the network. */

INVITE
F5
Proxy
1 ->
Proxy
2

INVITE sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length:132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F6
Proxy 1
-> A)
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0

INVITE F7
Proxy 2
-> B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>,<sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F8
Proxy 2
-> Proxy 1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F9
B -> Proxy 2

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0

180
Ringin
g F10
Proxy 2
->
Proxy
1

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F11
Proxy 1
->A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0

200 OK F12
B -> Proxy 2

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>,<sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK
F13
Proxy
2
->
Proxy
1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>,<sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F14
Proxy 1
-> A
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>,<sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F15
A -> Proxy 1

ACK sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@ss2.wcom.com>,<sip:UserB@there.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345601@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F16
Proxy 1 -> Proxy 2

ACK sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@there.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345601@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F17
Proxy 2 -> B

ACK sip: UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345601@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B */

/* User B Hangs Up with User A. */

BYE F18
User B
-> Proxy 2

BYE sip: UserA@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@ss1.wcom.com>,<sip:UserA@here.com>
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0

BYE
F19
Proxy
2 ->
Proxy
1

BYE sip: UserA@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0

BYE F20
Proxy 1 -> User A

BYE sip: UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0

200 OK F21
User A
-> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0

200 OK F22
Proxy 1 -> Proxy 2

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0
200 OK F23
Proxy 2 -> User B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0
3.1.2 Successful SIP to SIP with Proxy failure

In this scenario, User A completes a call to User B via a Proxy Server. User A is configured for a primary SIP Proxy Server SS1 and a secondary SIP Proxy Server SS2 (or is able to use DNS SRV records to locate SS1 and SS2). SS1 is out of service and does not respond to INVITEs (it is reachable, but unresponsive). After sending a CANCEL to SS1, User A then completes the call to User B using SS2.

Message Details

INVITE F1
A -> Proxy 1

INVITE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length:132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

INVITE F2
A -> Proxy 1

Same as Message F1

INVITE F3
A -> Proxy 1

Same as Message F1

INVITE F4
A -> Proxy 1
Same as Message F1

INVITE F5
A -> Proxy 1
Same as Message F1

INVITE F6
A -> Proxy 1
Same as Message F1

INVITE F7
A -> Proxy 1
Same as Message F1

/* User A gives up on the unresponsive proxy and sends a CANCEL. If any 200 OK responses come back to the INVITE, User A sends an ACK, then a BYE. */

CANCEL F8
A -> Proxy 1

CANCEL sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
INVITE F9
A -> Proxy 2

INVITE sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
/* Proxy 2 challenges User A for authentication */
407 Proxy Authorization Required F10
Proxy 2 -> User A

SIP/2.0 407 Proxy Authorization Required
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Proxy-Authenticate: Digest realm="MCI WorldCom SIP",
domain="wcom.com", nonce="ea9c8e88df84f1ce4341ae6cbe5a359",
opaque="", stale="FALSE", algorithm="MD5"
Content-Length: 0

ACK F11
A -> Proxy 2

ACK sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0

/* User A responds be re-sending the INVITE with authentication credentials in it. */

INVITE F12
A -> Proxy 2

INVITE sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345602@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP", nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="", url="sip:ss2.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length:132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Proxy 2 accepts the credentials and forwards the INVITE to User B. Client for A prepares to receive data on port 49170 from the network. */

INVITE F13
Proxy 2 ->B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345602@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length:132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
(100 Trying F14
Proxy 2 -> User A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 12345602@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F15
B -> Proxy 2

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345602@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F16
Proxy 2
-> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345602@here.com
CSeq: 1 INVITE
Content-Length: 0

200 OK F17
B -> Proxy 2

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345602@here.com
CSeq: 1 INVITE
SIP Telephony Call Flow Examples

Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F18
Proxy 2 -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345602@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 134

v=0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F19
A -> Proxy 2

ACK sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@there.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345602@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F20
Proxy 2 -> B
ACK sip: UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345602@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B */

/* User B Hangs Up with User A. */
BYE F21
User B
-> Proxy 2

BYE sip: UserA@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345602@here.com
CSeq: 1 BYE
Content-Length: 0

BYE F22
Proxy 2 -> User A

BYE sip: UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345602@here.com
CSeq: 1 BYE
Content-Length: 0

200 OK F23
User A
-> Proxy 2
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345602@here.com
CSeq: 1 BYE
Content-Length: 0

200 OK F24
Proxy 2 -> User B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345602@here.com
CSeq: 1 BYE
Content-Length: 0
3.1.3 Successful SIP to SIP through SIP Firewall Proxy

User A completes a call to User B through a Firewall Proxy and a SIP Proxy. The signaling message exchange is identical to 3.1.1 but the media stream setup is not end-to-end – the Firewall proxy terminates both media streams and bridges them. This is done by the Proxy modifying the SDP in the INVITE (F1) and 200 OK (F11) messages.

In addition to firewall traversal, this back-to-back User Agent Client and User Agent Server could be used as part of an anonymizer service (in which all identifying information on User A would be removed), or to perform codec media conversion, such as mu-law to A-law conversion of PCM on an international call.

Message Details

INVITE F1
A->SIP FW

INVITE sip:UserB@fwpl.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="",
uri="sip:ssl.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length:132

v=0
c=IN IP4 2890844526 2890844526 IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
Client for A prepares to receive data on port 49170 from the network.*/

INVITE F2
SS FW -> SS1

INVITE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@fwp1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization: Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cece4341ae6cbe5a359", opaque="",
uri="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length: 134

v=0
c=IN IP4 200.201.202.203
m=audio 1000 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F3
SIP FW-> A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* SIP FW prepares to proxy data from port 1000 to
100.101.102.103/49170. SS1 uses a location manager function to
determine where B is actually located. Based upon location analysis
the call is forwarded to User B */
INVITE F4
SS1->B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss1.wcom.com>,<sip:UserB@fwp1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length:134

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 200.201.202.203
m=audio 1000 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F5
SS1 -> SIP FW)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

(100 Trying F6
B -> SS1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
180 Ringing F7
B->SS1

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F8
SS1 -> SIP FW

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F9
SIP FW
-> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

200 OK F10
B->SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss1.wcom.com>,<sip:UserB@fwp1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 133

v=0
o=UserB 2890844527 2890844527 IN IP4 there.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F11
SS1 -> SIP FW

SIP/2.0 200 OK
Via: SIP/2.0/UDP gw1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ssl.wcom.com>,<sip:UserB@fwpl.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
o=UserB 2890844527 2890844527 IN IP4 there.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F12
SIP FW

-> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ssl.wcom.com>,<sip:UserB@fwpl.wcom.com>
From: TheBigGuy <sip:UserB@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
o=UserB 2890844527 2890844527 IN IP4 there.com
t=0 0
c=IN IP4 200.201.202.203
m=audio 1002 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* The gateway prepares to proxy packets from port 1001 to
110.111.112.113/3456
ACK F13
A->SIP FW

ACK sip:UserB@fwp1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:ss1.wcom.com>, <sip:UserB@there.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F14
SIP FW -> SS1

ACK sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@there.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F15
SS1->B

ACK sip: UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and the SIP GW and between the SIP GW and B*/

/* User A Hangs Up with User B */
BYE F16
A->SIP FW

BYE sip: UserB@fwp1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:ss1.wcom.com>, <sip:UserB@there.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

BYE F17
SIP FW
-> SS1

BYE sip: UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@there.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

BYE F18
SS1->B
BYE sip: UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F19
B->SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F20
SS1 -> SIP FW

SIP/2.0 200 OK
Via: SIP/2.0/UDP fwp1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F21
SIP FW
-> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
3.1.4
Successful SIP to SIP via Redirect and Proxy

In this scenario, User A places a call to User B using first a Redirect server then a Proxy Server. The INVITE message is first sent to the Redirect Server. The Server returns a 302 Moved Temporarily response (F2) containing a Contact header with User B’s current SIP address. User A then generates a new INVITE and sends to User B via the Proxy Server and the call proceeds normally.

The call is terminated when User B sends a BYE message.
Message Details

INVITE F1
A->Redir Proxy

INVITE sip:UserB@redirect.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length:132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/*Client for A prepares to receive data on port 49170 from the network.*/

302 Moved Temporarily F2
Redir Proxy
->A

SIP/2.0 302 Moved Temporarily
Contact: sip:UserB@ss2.wcom.com
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F3
A->Redir Proxy

ACK sip:UserB@redirect.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

INVITE F4
A -> Proxy

INVITE sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 2 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length:132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

INVITE F5
Proxy -> B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 2 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length:132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0

(100 Trying ) F7 B
-> Proxy

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0

180 Ringing F8
B->Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0

180 Ringing F9
Proxy->A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0

200 OK F10
B->Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
o=UserB 2890844527 2890844527 IN IP4 there.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F11
Proxy->A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
o=UserB 2890844527 2890844527 IN IP4 there.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F12
A -> Proxy
ACK sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@there.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 ACK
Content-Length: 0

ACK F13
Proxy -> B

ACK sip: UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 ACK
Content-Length: 0

/* RTP streams are established between A and B*/

/* User B Hangs Up with User A. */
BYE F14
B->Proxy

BYE sip: UserA@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: TheLittleGuy <sip:UserB@there.com>;tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0

BYE F15
Proxy->A
BYE sip: UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>; tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0

200 OK F16
A -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>; tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0

200 OK F17
Proxy -> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: TheLittleGuy <sip:UserB@there.com>; tag=314159
To: TheBigGuy <sip:UserA@here.com>
Call-Id: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0
3.2 Failure Scenarios

3.2.1 Unsuccessful SIP to SIP no answer

In this scenario, User A gives up on the call before User B answers (sends a 200 OK response). User A sends a CANCEL (F9) since no final response had been received from User B. If a 200 OK to the INVITE had crossed with the CANCEL, User A would have sent an ACK then a BYE to User B in order to properly terminate the call.

Note that the CSeq of the CANCEL message (F9) is not incremented. This is so that downstream clients can match the To, From, Call-ID, and CSeq of the CANCEL to the INVITE to decide which request to terminate.
Message Details

INVITE F1
A -> Proxy 1

INVITE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization: Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df8f1c3c4341ae6cbe5a359", opaque="",
uri="sip:ss1.wcom.com", response="dfe56131d195804f0689cd83306f7ecc"
Content-Type: application/sdp
Content-Length:132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/*Client for A prepares to receive data on port 49170 from the network.*/

INVITE F2
Proxy 1 -> Proxy 2

INVITE sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length:132

v=0
INVITE F4
Proxy 2 -> B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>,<sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F5
Proxy 2 -> Proxy 1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F6
B -> Proxy 2

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F7
Proxy 2 -> Proxy 1

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F8
Proxy 1
-> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
/* User A gives up and sends a CANCEL. If a 200 OK reply to the INVITE crossed with the CANCEL and was received by User A, User A would send an ACK then a BYE to terminate the call. */

CANCEL F9
A -> Proxy 1

CANCEL sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

CANCEL F10
Proxy 1 -> Proxy 2

CANCEL sip: UserA@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

CANCEL F11
Proxy 2 -> B

CANCEL sip: UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

200 OK F12
A -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

200 OK F13
Proxy 1 -> Proxy 2

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

200 OK F14
Proxy 2 ->B

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0
3.2.2
Unsuccessful SIP to SIP busy

In this scenario, User B is busy and sends a 486 Busy Here response to User A’s INVITE. The 4xx response is ACKed at each signaling leg.

Message Details

INVITE F1
User A
-> Proxy 1

INVITE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="",
uri="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/*Client for A prepares to receive data on port 49170 from
the network.*/
INVITE F2
Proxy 1 -> Proxy 2

INVITE sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F3
Proxy 1 -> User A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

INVITE F4
Proxy 2 ->
User B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>,<sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
(100 Trying F5
Proxy 2 -> Proxy 1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

486 Busy Here F6 User B
-> Proxy 2

SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F7
Proxy 2 -> User B

ACK sip: UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

486 Busy Here F8 Proxy 2 -> Proxy 1

SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
ACK F9
Proxy 1 -> Proxy 2

ACK sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

486 Busy Here F10
Proxy 1 -> User A

SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F11
User A
-> Proxy 1

ACK sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
3.2.3  Unsuccessful SIP to SIP no response

In this example, there is no response from User B to User A’s INVITE messages being re-transmitted by Proxy 2. After the sixth re-transmission, Proxy 2 gives up and sends a CANCEL to User B and a 480 No Response to User A. Note that the CANCEL would also be retransmitted six times, as governed by SIP timer T1 as in Section 5.2.6.

Message Details

INVITE F1
User A
-> Proxy 1

INVITE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c888df84f1cecc4341ae6cbe5a359", opaque="",
uri="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length: 132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
/*Client for A prepares to receive data on port 49170 from the network.*/
INVITE F2
Proxy 1 -> Proxy 2

INVITE sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F3
Proxy 1 -> User A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

INVITE F4
Proxy 2 -> User B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>,<sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F5
Proxy 2 -> Proxy 1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

INVITE F6
Proxy 2 -> User B

Resend of Message F4
INVITE F7
Proxy 2 -> User B

Resend of Message F4
INVITE F8
Proxy 2 -> User B

Resend of Message F4
INVITE F9
Proxy 2 -> User B

Resend of Message F4
INVITE F10
Proxy 2 -> User B

Resend of Message F4
INVITE F11
Proxy 2 -> User B

Resend of Message F4
CANCEL F12
Proxy 2 -> User B
CANCEL sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

480 No Response F13 Proxy 2 -> Proxy 1

SIP/2.0  480 No Response
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F14
Proxy 1 -> Proxy 2

ACK sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

480 No Response F15
Proxy 1 -> User A

SIP/2.0  480 No Response
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F16
User A
-> Proxy 1
ACK sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
3.2.4 Unsuccessful SIP to SIP Temporarily Unavailable

In this scenario, User B initially sends a 180 Ringing response to User A, indicating that alerting is taking place. However, then a 480 Unavailable is then sent to User A. This response is acknowledged then proxied back to User A.
Message Details

INVITE F1
A -> Proxy 1

INVITE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1ec4341ae6cbe5a359", opaque="",
uri="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
/*Client for A prepares to receive data on port 49170 from the
network.*/
INVITE F2
Proxy 1 -> Proxy 2

INVITE sip:UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
INVITE F3
Proxy 2 -> B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@ss2.wcom.com>,<sip:UserB@ss1.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4
Proxy 1 -> A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

(100 Trying F5
Proxy 2 -> Proxy 1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

(100 Trying F6
User B
-> Proxy 2)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F7
B -> Proxy 2

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F8
Proxy 2 -> Proxy 1

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F9
Proxy 1
-> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

480 Temporarily Unavailable F10
B -> Proxy 2

SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP ss2.wcom.com:5060
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F11
Proxy 2 ->B

ACK sip: UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

480 Temporarily Unavailable F12
Proxy 2 -> Proxy 1

SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
ACK F13
Proxy 1 -> Proxy 2

ACK sip: UserB@ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

480 Temporarily Unavailable F14
Proxy1
-> A

SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F15
A -> Proxy 1

ACK sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@there.com>;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
4 SIP to Gateway Dialing

In the following scenarios, User A (TheBigGuy sip:UserA@here.com) is a SIP phone or other SIP-enabled device. User B is reachable via the PSTN at global telephone number +1-972-555-2222. User A places a call to User B through a Proxy Server SS1 and a Network Gateway. In other scenarios, User A places calls to User C, who is served via a PBX (Private Branch Exchange) and is identified by a private extension 444-3333, or global number +1-918-555-3333. Note that User A uses his/her global telephone number +1-314-555-1111 in the From header in the INVITE messages. This then gives the Gateway the option of using this header to populate the calling party identification field in subsequent signaling (CgPN in ISUP). Left open is the issue of how the Gateway can determine the accuracy of the telephone number, necessary before passing it as a valid CgPN in the PSTN. Note that User A still uses his/her SIP URL in the Contact header.

There is a major SIP issue in the call flows in this section and Section 6. In-band alerting (ringing tone, busy tone, recorded announcements, etc.) is present in the PSTN speech path after the receipt of the SS7 Address Complete Message (ACM) which maps to the SIP 180 Ringing response. In a SIP to SIP call, the media path is not established until the call is answered (200 OK sent). In order for the SIP caller User A to hear this alerting, it is necessary that an early media path be established to perform this. This is the purpose of the 183 Session Progress[5] responses used throughout this document in place of the 180 Ringing.

This document will be updated as this issue is further refined, with the possible inclusion of reliable responses[6] and/or additional SIP headers.

4.1 Success Scenarios

In these scenarios, User A is a SIP phone or other SIP-enabled device. User A places a call to User B in the PSTN or User C on a PBX through a Proxy Server SS1 and a Gateway.
4.1.1 Successful SIP to ISUP PSTN call

User A dials the globalized E.164 number +1-972-555-2222 to reach User B. Note that A might have only dialed the last 7 digits, or some other dialing plan. It is assumed that the SIP User Agent Client converts the digits into a global number and puts them into a SIP URL.

User A could use either their SIP address (sip:UserA@here.com) or SIP telephone number (sip:+1-314-555-1111@ss1.wcom.com;user=phone) in the From header. In this example, the telephone number is included, and it is shown as being passed as calling party identification through the Network Gateway to User B (F5). Note that for this number to be passed into the SS7 network, it would have to be somehow verified for accuracy.

In this scenario, User B answers the call then User A disconnects the call. Signaling between NGW1 and User B’s telephone switch is SS7.

Message Details

INVITE F1  
A->SS1  
INVITE sip:+1-972-555-2222@ss1.wcom.com;user=phone  
SIP/2.0  
Via: SIP/2.0/UDP here.com:5060  
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone  
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone  
Call-Id: 123456000@here.com  
CSeq: 1 INVITE  
Contact: TheBigGuy <sip:UserA@here.com>  
Authorization:Digest username="UserA", realm="MCI WorldCom SIP", nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="", url="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"  
Content-Type: application/sdp  
Content-Length: 132

v=0  
c=IN IP4 here.com  
t=0  
m=audio 49170 RTP/AVP 0  
a=rtpmap:0 PCMU/8000
(100 Trying F2
SS1 -> User A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* SS1 uses a location manager function to determine where B is
actually located. Based upon location analysis the call is forwarded
to NGW1. Client for A prepares to receive data on port 49170 from
the network.*/

INVITE F3
SS1
-> NGW1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ss1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4
GW -> SS1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

IAM F5
GW -> User B

IAM
CdPN=972-555-2222,NPI=E.164,NOA=National
CgPN=314-555-1111,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
ACM F6
User B
-> GW

ACM
Charge Indicator=No Charge
Called Party Status=no indication
Called Party’s Category=ordinary subscriber
End To End Method=none available
Interworking=encountered
End to End Information=none available
ISUP Indicator=not used all the way
ISDN Access Terminating access non ISDN
Echo Control=not included
183 Session Progress F7
GW -> SS1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
/* SS1 proxies the OK to User A
At this point the GW will start sending an RTP path to the receive
port on A encoding anything that is being received from B via the
PSTN network (i.e. ringing) */

183 Session Progress F8
SS1
->User A

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ANM F9
User B
-> GW

ANM

200 OK F10
GW -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F11
SS1
->User A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F12
A->SS1

ACK sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:+1-972-555-2222@ngw1.wcom.com;user=phone>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
ACK F13
SS1 -> GW

ACK sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
 ;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B(via the GW) */

/* User A Hangs Up with User B */
BYE F14
A->SS1

BYE sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:+1-972-555-2222@ngw1.wcom.com;user=phone>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
 ;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

BYE F15
SS1 -> GW

BYE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
 ;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F16
GW -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F17
SS1->A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

REL F18
GW -> B

REL
CauseCode=16 Normal
CodingStandard=CCITT
RLC F19
B -> GW

RLC
4.1.2 Successful SIP to ISDN PBX call

User A is a SIP device while User C is connected via an Enterprise Gateway (GW1) to a PBX. The PBX connection is via an ISDN trunk group. User A dials User C’s telephone number (918-555-3333) which is globalized and put into a SIP URL.

The phone-context in the username portion of the Request-URI in message F3 is used to identify the context (customer, trunk group, or line) in which the private number 444-3333 is valid. Otherwise, this INVITE message could get forwarded and the context of the digits could become lost and the call unroutable. See section 1.1 for a discussion of phone-context.

Proxy SS1 looks up the telephone number and locates the Enterprise Gateway that servers User C. User C is identified by its extension (444-3333) in the Request-URI sent to GW1.

User A hears the ringing provided by the Gateway on the media path established after the 183 Session Progress response is received. Signaling between GW1 and PBX C is shown as ISDN.

Message Details

INVITE F1
A->SS1
INVITE sip:+1-918-555-3333@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="",
uri="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 here.com
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to GW1 with the extension determined as 444-3333. Client for A prepares to receive data on port 49170 from the network. */

INVITE F3
SS1 -> GW1

INVITE sip:444-3333,phone-context=p1234@gw1.wcom.com;user=phone
SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
o= UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4
GW -> SS1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

SETUP F5
GW -> User C

Protocol discriminator=Q.931
Call reference: Flag=0, CR value=any valid value not in use
Message type=SETUP
Bearer capability: Information transfer capability=0 (Speech) or 16
(3.1 kHz audio)
Channel identification=Preferred or exclusive B-channel
Progress indicator=1 (Call is not end-to-end ISDN; further call
progress information may be available inband)
Called party number:
Type of number and numbering plan ID=?? (private numbering plan)
Digits=444-3333
CALL PROCeeding F6
User C
-> GW

Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F5 SETUP message
Message type=CALL PROC
Channel identification=Exclusive B-channel
PROGress F7
User C
-> GW

Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F5 SETUP message
Message type=PROG
Progress indicator=1 (Call is not end-to-end ISDN; further call
progress information may be available inband)
183 Session Progress F8
GW -> SS1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-318-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* The GW will establish an RTP path to the receive port on A encoding anything that is being received from C via the PSTN network (i.e. ringing)
183 Session Progress F9 SS1
->User A

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

CONNect F10
User C
-> GW

Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F5 SETUP message
Message type=CONN
CONNect ACK F11
GW -> User C

Protocol discriminator=Q.931
Call reference: Flag=0, CR value=value in F5 SETUP message
Message type=CONN ACK
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: sip:444-3333,phone-context=p1234@gw1.wcom.com ;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: sip:444-3333,phone-context=p1234@gw1.wcom.com ;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
ACK sip:444-3333,phone-context=p1234@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:444-3333,phone-context=p1234@gw1.wcom.com> ;user=phone
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F15
SS1 -> GW

ACK sip:444-3333,phone-context=p1234@gw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B(via the GW) */

/* User A Hangs Up with User B. */
BYE F16
A->SS1

BYE sip:444-3333,phone-context=p1234@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:444-3333,phone-context=p1234@gw1.wcom.com>;user=phone
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
BYE F17
SS1 -> GW

BYE sip:444-3333,phone-context=p1234@gw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F18
GW -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F19
SS1->A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheOtherGuy <sip:+1-918-555-3333@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

DISConnect F20
GW -> User C

Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F4 SETUP message
Message type=DISC
Cause=16 (Normal clearing)
RELease F21
User C
  -> GW

Protocol discriminator=Q.931
Call reference: Flag=0, CR value=value in F4 SETUP message
Message type=REL
RELease COMplete F22
GW -> User C

Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F4 SETUP message
Message type=REL COM
4.1.3
Successful SIP to ISUP PSTN call with overflow
User A calls User B through SS1 working as a proxy server. SS1 tries an Enterprise Gateway GW1. GW1 is not available and responds with a 503 Service Unavailable (F4). The call is then routed to a Network Gateway NGW2. User B answers the call. The call is terminated when User A disconnects the call. NGW2 and User B’s telephone switch use SS7 signaling.

Message Details

INVITE F1
A->SS1

INVITE sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1ce4341ae6cbe5a359", opaque="",
url="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length: 132

v=0
o= UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c= IN IP4 here.com
m= audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS1 uses a location manager function to determine where B is actually located. SS1 receives a primary route NGW1 and a secondary route NGW2. NGW1 is tried first */

INVITE F2
SS1 -> NGW1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP sgw1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F3 Proxy -> User A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

503 Service Unavailable F4
NGW1-> SS1

SIP/2.0 503 Service Unavailable
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=123456789
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* See table below for possible responses resulting in attempt to
next route in list.

408 Request Timeout
413 Request Entity Too Large
480 Temporarily Unavailable
500 Server Internal Error
501 Not Implemented
502 Bad Gateway
503 Service Unavailable
504 Gateway Timeout
505 Version Not Supported
*

ACK F5
SS1 -> GW2

ACK sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
 ;user=phone;tag=123456789
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* SS1 now tries secondary route to NGW2 */
INVITE F6 SS1
-> NGW2

INVITE sip:+1-972-555-2222@ngw2.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
IAM F7
NGW2 -> User B

IAM
CdPN=972-555-2222,NPI=E.164,NOA=National
CgPN=314-555-1111,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required

ACM F8
User B
-> NGW2

ACM
Charge Indicator=No Charge
Called Party Status=no indication
Called Party’s Category=ordinary subscriber
End To End Method=none available
Interworking=encountered
End to End Information=none available
ISUP Indicator=not used all the way
ISDN Access Terminating access non ISDN
Echo Control=not included

183 Session Progress F9
NGW2
-> SS1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
/* The GW will establish an RTP path to the receive port on A 
encoding anything that is being received from B via the PSTN network 
(i.e. ringing) */

183 Session Progress F10
SS1 -> User A

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ANM F11
User B
-> NGW2

ANM

200 OK F12
NGW2
-> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ssl1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ssl1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ssl1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: sip:+1-972-555-2222@ngw2.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F13
SS1
-> User A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: sip:+1-972-555-2222@ngw2.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F14
A->SS1

ACK sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:+1-972-555-2222@ngw2.wcom.com>;user=phone
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F15
SS1 -> NGW2
ACK sip:+1-972-555-2222@ngw2.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ssl.wcom.com>
;user=phone;tag=314159
Call-Id: 123456000@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B(via the GW) */

/* User A Hangs Up with User B. */
BYE F16
A->SS1

BYE sip:+1-972-555-2222@ssl.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:+1-972-555-2222@ngw2.wcom.com>;user=phone
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ssl.wcom.com>
;user=phone;tag=314159
Call-Id: 123456000@here.com
CSeq: 2 BYE
Content-Length: 0

BYE F17
SS1 --> NGW2

BYE sip:+1-972-555-2222@ngw2.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ssl.wcom.com>
;user=phone;tag=314159
Call-Id: 123456000@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F18
NGW2 --> SS1
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F19
SS1-> User A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

REL F20
GW -> B

REL
CauseCode=16 Normal
CodingStandard=CCITT
RLC F21
B -> GW

RLC
4.2 Failure Scenarios

In these failure scenarios, the call does not complete. In most cases, however, a media stream is still setup. This is due to the fact that most failures in dialing to the PSTN result in in-band tones (busy, reorder tones) or announcements ("The number you have dialed has changed. The new number is..."). The 183 Session Progress[5] response containing SDP media information is used to setup this early media path so that the caller User A knows the final disposition of the call.

The media stream is either terminated by the caller after the tone or announcement has been heard and understood, or by the Gateway after a timer expires.

In other failure scenarios, a SS7 Release with Cause Code is mapped to a SIP response. In these scenarios, the early media path is not used, but the actual failure code is conveyed to the caller by the SIP User Agent Client.
4.2.1
Unsuccessful SIP to PSTN call: Treatment from PSTN

User A calls User B in the PSTN through a proxy server SS1 and a Network Gateway NGW1. The call is rejected by the PSTN with an in-band treatment (tone or recording) played. User A hears the treatment and then issues a CANCEL (F9) to terminate the call. (A BYE is not sent since no final response was ever received by User A.)

Message Details

INVITE F1
A→SS1

INVITE sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="",
uri="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length: 132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F2 SS1 → A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to NGW1. Client for A prepares to receive data on port 49170 from the network.*/

INVITE F3
SS1 -> NGW1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 123456000@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 here.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4 NGW1-> SS1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 123456000@here.com
CSeq: 1 INVITE
Content-Length: 0

IAM F5
GW -> User B

IAM
CdPN=972-555-2222,NPI=E.164,NOA=National
CgPN=314-555-1111,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
ACM F6
User B
-> GW

ACM
Charge Indicator=No Charge
Called Party Status=no indication
Called Party's Category=ordinary subscriber
End To End Method=none available
Interworking=encountered
End to End Information=none available
ISUP Indicator=not used all the way
ISDN Access Terminating access non ISDN
Echo Control=not included
183 Session Progress F7
GW -> SS1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ssl1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ssl1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

183 Session Progress F8
SS1
-> User A

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ssl1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150
/* User A listens to recorded announcement from the PSTN then hangs up */

CANCEL F9
A->SS1

CANCEL sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

CANCEL F10
SS1 -> GW

CANCEL sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

200 OK F11
GW -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

200 OK F12
SS1->A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

REL F13
GW -> B

REL
CauseCode=16 Normal
CodingStandard=CCITT
RLC F14
B -> GW

RLC
4.2.2

Unsuccessful SIP to PSTN: REL w/Cause from PSTN

User A calls PSTN User B through a Proxy Server SS1 and a Network Gateway NGW1. However, User A does not provide enough digits for the call to be completed. (In a real scenario, this call might have been rejected by SS1 based on incomplete address. However, especially on international calls, the number of digits in the number is not obvious, and this scenario may result.) The call is rejected by the PSTN with a SS7 Release message REL containing a specific Cause value. This cause value (28) is mapped by the Gateway to a SIP 484 Address Incomplete response which is proxied back to User A.

Message Details

INVITE F1
A->SS1

INVITE sip:+1-972-555-222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="",
uri="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length: 132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying  F2 SS1 -> A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to NGW1. Client for A prepares to receive data on port 49170 from the network. */

INVITE F3
SS1
-> NGW1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 123456000@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
t=0 0
c=IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4 NGW1-> SS1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 123456000@here.com
CSeq: 1 INVITE
Content-Length: 0

IAM F5
GW -> User B

IAM
CdPN=972-555-2222,NPI=E.164,NOA=National
CgPN=314-555-1111,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI -Not Required
REL F6
User B
   -> GW

REL
CauseValue=28 Address Incomplete
CodingStandard=CCITT
RLC F7
GW -> User B

RLC

/* Network Gateway maps CauseValue=28 to the SIP message 484 Address Incomplete */

484 Address Incomplete F8
GW -> SS1

SIP/2.0 484 Address Incomplete
Via: SIP/2.0/UDP ssl.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-222@ss1.wcom.com>;user=phone;
tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F9
SS1 -> GW

ACK sip:+1-972-555-222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-222@ss1.wcom.com>;user=phone;
tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

484 Address Incomplete F10
SS1 -> User A
SIP/2.0 484 Address Incomplete
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-222@ss1.wcom.com>;user=phone;
tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F11
User A
   -> SS1

ACK sip:+1-972-555-222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-222@ss1.wcom.com>;user=phone;
tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
4.2.3  Unsuccessful SIP to PSTN: ANM Timeout

User A calls User B in the PSTN through a proxy server SS1 and a Gateway GW1. The call is released by the Gateway after its ISUP T9 timer expires due to no ANswer Message (ANM) being received. The Gateway sends a SS7 Release REL message to the PSTN and a 480 Temporarily Unavailable response to User A in the SIP network.
Message Details

INVITE F1
A->SS1

INVITE sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Authorization:Digest username="UserA", realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="",
uri="sip:ss1.wcom.com", response="dfe56131d1958046689cd83306477ecc"
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 here.com
t=0 0
c=IN IP4 here.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS1 uses a location manager function to determine where B is
actually located. Based upon location analysis the call is forwarded
to GW1. Client for A prepares to receive data on port 49170 from the
network.*/

(100 Trying F2 SS1
 -> A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

INVITE F3
SS1 -> GW1
INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-314-555-1111@ss1.wcom.com>
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 132

v=0
c=IN IP4 here.com
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4 GW1
  -> SS1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

IAM F5
GW --> User B

IAM
CdPN=972-555-2222,NPI=E.164,NOA=National
CgPN=314-555-1111,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
ACM F6
User B
  --> GW

ACM
Charge Indicator=No Charge
Called Party Status=no indication
Called Party’s Category=ordinary subscriber
End To End Method=none available
Interworking=encountered
End to End Information=none available
ISUP Indicator=not used all the way
ISDN Access Terminating access non ISDN
Echo Control=not included
183 Session Progress F7
GW -> SS1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ssl1.wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ssl1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

183 Session Progress F8
SS1 -> User A

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ssl1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
/* After ISUP T9 Timer expires, Network Gateway sends REL to ISUP network and 480 to SIP network */

REL F9 GW -> User B
REL
  CauseCode=16 Normal
  CodingStandard=CCITT
  RLC F10 User B
    -> GW
RLC
  480 Temporarily Unavailable F11
GW -> SS1
SIP/2.0 480 Temporarily Unavailable
  Via: SIP/2.0/UDP ssl.wcom.com:5060
  Via: SIP/2.0/UDP here.com:5060
  From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
  To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
    ;user=phone;tag=314159
  Call-Id: 12345600@here.com
  CSeq: 1 INVITE
  Content-Length: 0

ACK F12
SS1 -> GW
ACK sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
  Via: SIP/2.0/UDP ssl.wcom.com:5060
  From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
  To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
    ;user=phone;tag=314159
  Call-Id: 12345600@here.com
  CSeq: 1 ACK
  Content-Length: 0

480 Temporarily Unavailable F13
SS1 -> User A
SIP/2.0 480 Temporarily Unavailable
  Via: SIP/2.0/UDP here.com:5060
  From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
  To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
    ;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F14
User A
-> SS1

ACK sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
To: TheLittleGuy <sip:+1-972-555-2222@ss1.wcom.com>
;user=phone;tag=314159
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
5 Gateway to SIP Dialing

5.1 Success Scenarios

In these scenarios, User A is placing calls from the PSTN to User B in a SIP network. User A’s telephone switch signals to a Network Gateway (NGW1) using SS7.

Since the called SIP User Agent does not send in-band signaling information, no early media path needs to be established on the IP side. As a result, the 183 Session Progress response is not used. However, NGW1 will establish a one way speech path prior to call completion, and generate ringing for the PSTN caller. Any tones or recordings are generated by NGW1 and played in this speech path. When the call completes successfully, NGW1 bridges the PSTN speech path with the IP media path.
5.1.1 Successful PSTN to SIP call

In this scenario, User A from the PSTN calls User B through a Network Gateway NGW1 and Proxy Server SS1. When User B answers the call the media path is setup end-to-end. The call terminates when User A hangs up the call, with User A’s telephone switch sending a SS7 Release message which is mapped to a BYE by NGW1.

Message Details

IAM F1
User A -> GW

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
INVITE F2
A->SS1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to GW1. GW1 prepares to receive data on port 3456 from User A.*/
INVITE F3
SS1
-> User B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4 User B -> SS1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F5 User B -> SS1

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F6 SS1
NGW1 ->

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

ACM F7
NGW1 -> User A

ACM
Charge Indicator=No Charge
Called Party Status=no indication
Called Party’s Category=ordinary subscriber
End To End Method=none available
Interworking=encountered
End to End Information=none available
ISUP Indicator=not used all the way
ISDN Access Terminating access non ISDN
Echo Control=not included
200 OK F8 User B -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
Contact: TheLittleGuy <sip:UserB@there.com>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F9 SS1
-> NGW1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F10 GW1 -> SS1

ACK sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:UserB@there.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

ACK F11 SS1 -> User B

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

ANM F12
User B
-> NGW1

ANM
/* RTP streams are established between A and B (via the GW) */

/* User A Hangs Up with User B. */
REL F13
User A
    -> NGW1

REL
CauseCode=16 Normal
CodingStandard=CCITT
RLC F14
NGW1-> User A

RLC
BYE F15
NGW1-> SS1

BYE sip:UserB@ssl.wcom.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:UserB@there.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0

BYE F16
SS1 -> User B

BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ssl1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0

200 OK F17
User B
    -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ssl1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0

200 OK F18 SS1 -> NGW1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0
5.1.2 Successful PSTN to SIP call, Fast Answer

This "fast answer" scenario is similar to 5.1.1 except that User B immediately accepts the call, sending a 200 OK (F5) without sending a 180 Ringing response. The Gateway then sends an Answer Message (ANM) without sending an Address Complete Message (ACM).

Message Details

IAM F1
User A -> GW

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
INVITE F2
GW -> SS1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to User B. User B prepares to receive data on port 3456 from User A.*/
INVITE F3
SS1 --> User B

INVITE UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4
SS1 --> GW1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

200 OK F5 User B --> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 150
v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F6 SS1
-> GW1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F7 GW1
-> SS1

ACK UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:UserB@there.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

ACK F8 SS1
-> User B

ACK UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

ANM F9
User B
->GW

ANM

/* RTP streams are established between A and B (via the GW) */

/* User A Hangs Up with User B */
REL F10
User A
->GW

REL
CauseCode=16 Normal
CodingStandard=CCITT
RLC F11
GW -> User A

RLC
BYE F12
GW -> SS1

BYE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:UserB@there.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0

BYE F13
SS1 -> User B

BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0

200 OK F14
User B
-> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ssl1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0

200 OK F15 SS1 ->GW

SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0
5.1.3 Successful PBX to SIP call

In this scenario, User A calls from PBX A to User B through GW1 and SS1 working as a proxy server. Signaling between PBX A and GW1 is Feature Group B (FGB) circuit associated signaling (in-band mult-frequency outpulsing). After the receipt of the 180 Ringing from User B, GW1 generates ringing tone for User A.

User B answers the call by sending a 200 OK. The call terminates when User A hangs up, causing GW1 to send a BYE.

Message Details

MF Digits F1
PBX A
->GW1

KP 1 972 555 2222 ST
INVITE F2
A->SS1

INVITE sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone
Call-Id: 12345602@gw1.wcom.com
CSeq: 1 INVITE
Contact: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to GW1.*/
INVITE F3
SS1
-> User B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone
Call-Id: 12345602@gw1.wcom.com
CSeq: 1 INVITE
Contact: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4 SS1
-> GW)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@gw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F5 User B -> SS1

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone
;tag=314159
Call-Id: 12345602@gw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F6 SS1
-> GW1

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@gw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

/* One way Voice path is established between GW and the PBX for ringing. */

200 OK F7 User B -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@gw1.wcom.com
Contact: TheLittleGuy <sip:UserB@there.com>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 134

v=0
o=UserB 2890844527 2890844527 IN IP4 there.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F8 SS1
-> GW1

SIP/2.0 200 OK
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@gw1.wcom.com
CSeq: 1 INVITE
Contact: TheLittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
o=UserB 2890844527 2890844527 IN IP4 there.com
t=0 0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F9 GW1
->SS1

ACK sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP gw1.wcom.com:5060
Route: <sip:UserB@there.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@gw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

ACK F10 SS1
->User B

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B (via the GW) */

/* User A Hangs Up with User B. */
BYE F11
GW -> SS1

BYE sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP gw1.wcom.com:5060
Route: <sip:UserB@there.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@gw1.wcom.com
CSeq: 2 BYE
Content-Length: 0

BYE F12
SS1 -> User B

BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@gw1.wcom.com
CSeq: 2 BYE
Content-Length: 0

200 OK F13
User B
-> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0

200 OK F14 SS1 ->GW

SIP/2.0 200 OK
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>;user=phone
To: sip:+1-972-555-2222@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@gw1.wcom.com
CSeq: 2 BYE
Content-Length: 0
5.2 Failure Scenarios

5.2.1 Unsuccessful PSTN to SIP REL, SIP error mapped to REL

User A attempts to call a SIP user through SS1 working as a proxy server. SS1 is unable to find any routing for the number. The call is rejected by SS1 with a REL message containing a specific Cause value mapped by the gateway based on the SIP error.

Message Details

IAM F1
User A
-> GW

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-9999,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
INVITE F2
A->SS1

INVITE sip:+1-972-555-9999@ngw1.wcom.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-9999@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to GW1. GW1 prepares to receive data on port 3456 from User A. */

604 Does Not Exist Anywhere F3
SS1 -> GW

SIP/2.0 604 Does Not Exist Anywhere
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-9999@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

ACK F4 GW1
-> SS1

ACK sip:+1-972-555-9999@ssl.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-9999@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

REL F5
GW -> User A

REL
CauseCode=1
CodingStandard=CCITT
RLC F6
User A
-> GW

RLC
5.2.2 Unsuccessful PSTN to SIP REL, SIP busy mapped to REL

In this scenario, User A calls User B through a Gateway GW1 and SS1 working as a proxy server. The call is routed to User B via the gateway. The call is rejected by the User B who sends a 600 Busy Everywhere response. The Gateway sends a REL message containing a specific Cause value mapped by the gateway based on the SIP error.

Since no interworking is indicated in the IAM (F1), the busy tone is generated locally by User A’s telephone switch. In scenario 5.2.3, the busy signal is generated by the Gateway since interworking is indicated.
Message Details

IAM F1
User A
-> GW

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
INVITE F2
A->SS1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to GW1. GW1 prepares to receive data on port 3456 from User A.*/

INVITE F3
SS1
->User B

INVITE UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ssl1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4 SS1
-> GW1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

600 Busy Everywhere F5 User B
-> SS1

SIP/2.0 600 Busy Everywhere
Via: SIP/2.0/UDP ssl1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

ACK F6 SS1
-> User B

ACK UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ssl1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

600 Busy Everywhere F7 SS1 -> GW1

SIP/2.0 600 Busy Everywhere
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

ACK F8 GW1
-> SS1

ACK UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

REL F9
User A
-> GW

REL
CauseCode=17 Busy
CodingStandard=CCITT
RLC F10
GW -> User A

RLC
5.2.3
Unsuccessful PSTN->SIP, SIP error interworking to tones

In this scenario, User A calls User B through SS1 working as a proxy server. The call is routed to User B via the gateway. The call is rejected by the User B client. GW1 plays busy tone, and releases call after timeout.

GW1 plays the busy tone since the IAM (F1) indicates the interworking is present. In scenario 5.2.2, with no interworking, the busy indication is carried in the REL Cause value and is generated locally instead.
Message Details

IAM F1
User A
-> GW

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
Interworking=encountered
INVITE F2
A->SS1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 289084527 289084527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS1 uses a location manager function to determine where B is
actually located.  Based upon location analysis the call is forwarded
to GW1.  GW1 prepares to receive data on port 3456 from User A.*/

INVITE F3
SS1
-> User B

INVITE UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4 User B -> SS1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

600 Busy Everywhere F5 User B
-> SS1

SIP/2.0 600 Busy Everywhere
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

ACK F6 SS1
-> User B

ACK UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

600 Busy Everywhere F7 SS1 -> GW1

SIP/2.0 600 Busy Everywhere
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

ACK F8 GW1
->SS1

ACK sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

ACM F9
User B
->GW

ACM
Charge Indicator=No Charge
Called Party Status=no indication
Called Party’s Category=ordinary subscriber
End To End Method=none available
Interworking=encountered
End to End Information=none available
ISUP Indicator=not used all the way
ISDN Access Terminating access non ISDN
Echo Control=not included

/* One way speech path established between GW and User A */
/* Call Released after NGW treatment timer expires. */

REL F10
User A
->GW

REL
CauseCode=17
CodingStandard=CCITT
RLC F11
GW -> User A

RLC
5.2.4
Unsuccessful PSTN->SIP, ACM timeout

User A calls User B through SS1 working as a proxy server. SS1 re-
sends the INVITE after the expiration of SIP timer T1. User B never
responds with 180 Ringing _ it is reachable but unresponsive. After
the expiration of ISUP T7 timer, User A’s network disconnects the
call by sending a Release message REL. The Gateway maps this to a
CANCEL which is re-sent by SS1 after SIP T1 timer expires.

Message Details

IAM F1
User A
-> GW

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
INVITE F2
A->SS1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to GW1. GW1 prepares to receive data on port 3456 from User A.*/

INVITE F3
SS1
-> User B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 289084527 289084527 IN IP4 gatewayone.wcom.com
ct=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

100 Trying F4 SS1
-> GW

SIP/2.0 100 Trying
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

INVITE F5
GW -> SS1

Same as Message F3

INVITE F6
SS1
-->User B
Same as Message F3

INVITE F7
SS1
-->User B
Same as Message F3

INVITE F8
SS1
-->User B
Same as Message F3

INVITE F9
SS1
-->User B
Same as Message F3

/* ISUP Timer T7 expires in User A’s access network. */

REL F10
User A
-->GW

REL
CauseCode=16 Normal
CodingStandard=CCITT
RLC F11
GW --> User A

RLC
CANCEL F12
GW --> SS1

CANCEL sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0

CANCEL F13
SS1 -> User B

CANCEL sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ssl1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0

CANCEL F14
GW -> SS1

Same as Message F13

CANCEL F15
SS1
-> User B

Same as Message F13

CANCEL F16
SS1
-> User B

Same as Message F13

CANCEL F17
SS1
-> User B

Same as Message F13

CANCEL F18
SS1
->User B

Same as Message F13
5.2.5
Unsuccessful PSTN->SIP, ACM timeout, stateless SPS

In this scenario, User A calls User B through SS1 working as a stateless proxy server. Since SS1 is stateless, GW1 re-sends the INVITE and CANCEL messages after the expiration of SIP timer T1. User B does not respond with 180 Ringing. User A’s network disconnects the call with a release REL.
Message Details

IAM F1
User A
-> GW

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
INVITE F2
GW -> SS1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to GW1. GW1 prepares to receive data on port 3456 from User A.*/

INVITE F3
SS1-> User B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

INVITE F4
GW -> SS1

Same as Message F2

INVITE F5
SS1 -> User B

Same as Message F3

INVITE F6
GW -> SS1

Same as Message F2

INVITE F7
SS1 -> User B

Same as Message F3

INVITE F8
GW -> SS1

Same as Message F2

INVITE F9
SS1 -> User B

Same as Message F3
INVITE F10
GW -> SS1
Same as Message F2

INVITE F11
SS1 -> User B
Same as Message F3

INVITE F12
GW -> SS1
Same as Message F2

INVITE F13
SS1 -> User B
Same as Message F3

/* ISUP T7 Timer expires in User A’s access network. */

REL F14
User A
->GW
REL
CauseCode=16 Normal
CodingStandard=CCITT
RLC F15
GW -> User A

RLC
CANCEL F16
GW -> SS1

CANCEL sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0

CANCEL F17
SS1 -> User B

CANCEL sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0

CANCEL F18
GW -> SS1

Same as Message F16
CANCEL F19
SS1 -> User B

Same as Message F17
CANCEL F20
GW -> SS1

Same as Message F16
CANCEL F21
SS1 -> User B

Same as Message F17
CANCEL F22
GW -> SS1

Same as Message F16
CANCEL F23
SS1 -> User B

Same as Message F17
CANCEL F24
GW -> SS1

Same as Message F16
CANCEL F25
SS1 -> User B

Same as Message F17
CANCEL F26
GW -> SS1

Same as Message F16
CANCEL F27
SS1 -> User B

Same as Message F17
5.2.6
Unsuccessful PSTN->SIP, ANM timeout

In this scenario, User A calls User B through SS1 working as a proxy server. User B does not respond with 200 OK. User A disconnects the call with a Release message REL which is mapped by GW1 to a CANCEL. Note that if User B had sent a 200 OK response after the REL, GW1 would have sent an ACK then a BYE to properly terminate the call.
Message Details

IAM F1
User A
--> GW

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required

INVITE F2
A-->SS1

INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
c=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS1 uses a location manager function to determine where B is actually located. Based upon location analysis the call is forwarded to GW1. GW1 prepares to receive data on port 3456 from User A.*/

INVITE F3
SS1
-->User B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 150

v=0
o=GATEWAY1 2890844527 2890844527 IN IP4 gatewayone.wcom.com
t=0 0
c=IN IP4 gatewayone.wcom.com
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F4 User B -> SS1)
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F5 User B -> SS1
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing F6 SS1
-> GW1
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

ACM F7
GW -> User A

ACM
Charge Indicator=No Charge
Called Party Status=no indication
Called Party’s Category=ordinary subscriber
End To End Method=none available
Interworking=encountered
End to End Information=none available
ISUP Indicator=not used all the way
ISDN Access Terminating access non ISDN
Echo Control=not included

/* ISUP Timer T9 expires in User A’s access network. */

REL F8
User A
-> GW

REL
CauseCode=16 Normal
CodingStandard=CCITT
RLC F9
GW -> User A

RLC
CANCEL F10
GW -> SS1

CANCEL sip:UserB@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0

CANCEL F11
SS1 -> User B
CANCEL sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0

200 OK F12
User B
-> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0

200 OK F13 SS1 -> GW

SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-972-555-2222@ngw1.wcom.com;user=phone;tag=314159
Call-Id: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0
6 Gateway to Gateway Dialing via SIP Network

In these scenarios, both the caller and the called party are in the telephone network, either normal PSTN subscribers or PBX extensions. The calls route through two Gateways and at least one SIP Proxy Server. The Proxy Server performs the authentication and location of the Gateways.

Note that the proposed INFO method[8] is not currently included in this document. It is anticipated that a future version will include an example of this.

Again it is noted that the intent of this call flows document is not to provide a detailed parameter level mapping of SIP to PSTN protocols. For information on SIP to ISUP mapping, the reader is referred to other references[9].

6.1 Success Scenarios

In these scenarios, the call is successfully completed between the two Gateways allowing the PSTN or PBX users to communicate. The 183 Session Progress response is used to establish a media path between the two Gateways, allowing in-band alerting to pass from the called party telephone switch to the caller.
6.1.1 Successful ISUP PSTN to ISUP PSTN call

In this scenario, User A in the PSTN calls User C who is served as an extension on a PBX. User A’s telephone switch signals via SS7 to the Network Gateway NGW1, while User C’s PBX signals via SS7 with the Enterprise Gateway GW2. The CdPN and CgPN are mapped into SIP URLs and placed in the To and From headers. SS1 looks up the dialed digits in the Request-URI and maps the digits to the PBX extension of User C served by GW2. The INVITE is then forwarded to GW2 for call completion. An early media path is established end-to-end so that User A can hear the ringing tone generated by PBX C.

User C answers the call and the media path is cut through in both directions. User B hangs up terminating the call.

Message Details

IAM F1
User A
-> GW

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
INVITE F2
GW1 -> SS1

INVITE sip:+1-918-555-3333@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-918-555-3333@ss1.wcom.com;user=phone
Call-Id: 12345600@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 134

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
/ * SS uses a location manager function to determine where B is actually located. Response is returned listing onnet and offnet routes. */

INVITE F3
SS1 -> GW2

INVITE sip:444-3333,phone-context=p1234@gw2.wcom.com;user=phone
SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-918-555-3333@ss1.wcom.com;user=phone
Call-Id: 12345600@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 134

v=0
c=GW1 2890844526 2890844526 IN IP4 gw1.wcom.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

IAM F4
GW2 -> User C

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=444-3333,NPI=Private,NOA=Subscriber
USI=Speech
CPT=0 0
C=Normal
CCI =Not Required
ACM F5
User C
-> GW2

ACM
Charge Indicator=No Charge
Called Party Status=no indication
Called Party's Category=ordinary subscriber
End To End Method=None available
Interworking=encountered
End to End Information=None available
ISUP Indicator=not used all the way
ISDN Access Terminating access non ISDN
Echo Control=not included

/* Based on PROGress message, GW3 returns a 183 response with SDP allowing in-band call progress indications to be sent to the originator. */

183 Session Progress F6
GW2 -> SS1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345600@ngw1.wcom.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 134

v=0
o=PBX_B 987654321 987654321 IN IP4 gw2.wcom.com
t=0 0
c=IN IP4 100.101.102.104
m=audio 14918 RTP/AVP 0
a=rtpmap:0 PCMU/8000

183 Session Progress F7
SS1 -> GW1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345600@ngw1.wcom.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 134

v=0
o=PBX_B 987654321 987654321 IN IP4 gw2.wcom.com
t=0 0
c=IN IP4 100.101.102.104
m=audio 14918 RTP/AVP 0
da=rtpmap:0 PCMU/8000

/* GW1 receives packets from GW2 with encoded ringback, tones or other audio. GW1 decodes this and places it on the originating trunk. */

ACM F8
GW1
-> User A

ACM
Charge Indicator=No Charge
Called Party Status=no indication
Called Party's Category=ordinary subscriber
End To End Method=none available
Interworking=encountered
End to End Information=none available
ISUP Indicator=not used all the way
ISDN Access Terminating access non ISDN
Echo Control=not included

/* User B answers */
ANM F9
User C
-> GW2

ANM

200 OK F10
GW2 -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345600@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:444-3333,phone-context=p1234@gw2.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 134

v=0
200 OK F11
SS1 -> GW1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com>
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345600@ngw1.wcom.com
CSeq: 1 INVITE
Contact: sip:444-3333,phone-context=p1234@gw2.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 134

v=0
c=IN IP4 100.101.102.104
m=audio 14918 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ANM F12
GW1 -> User A

ANM

ACK F13
GW1 -> SS1

ACK sip:444-3333,phone-context=p1234@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:444-3333,phone-context=p1234@gw2.wcom.com>;user=phone
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

ACK F14
SS1 -> GW3
ACK sip:444-3333,phone-context=p1234@gw2.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
To: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
Call-Id: 12345600@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between GW1 and GW2. */

/* User B Hangs Up with User A. */
REL F15
User C-> GW2

REL
CauseCode=16 Normal
CodingStandard=CCITT
BYE F16
GW3 -> SS1

BYE sip:+1-314-555-1111@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP gw2.wcom.com:5060
Route: <sip:+1-314-555-1111@ngw1.wcom.com>;user=phone
From: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
To: sip:+1-314-555-1111@ngw1.wcom.com;user=phone
Call-Id: 12345600@ngw1.wcom.com
CSeq: 4 BYE
Content-Length: 0

RLC F17
GW2
->User C

RLC
BYE F18
SS1 -> GW1

BYE sip:+1-314-555-1111@gw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw2.wcom.com:5060
From: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
To: sip:+1-314-555-1111@ngw1.wcom.com
Call-Id: 12345600@ngw1.wcom.com
CSeq: 4 BYE
Content-Length: 0

200 OK F19
GW1 -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw2.wcom.com:5060
From: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
To: sip:+1-314-555-1111@ngw1.wcom.com
Call-Id: 12345600@ngw1.wcom.com
CSeq: 4 BYE
Content-Length: 0

200 OK F20
SS11
->GW3

SIP/2.0 200 OK
Via: SIP/2.0/UDP gw2.wcom.com:5060
From: sip:+1-918-555-3333@ss1.wcom.com;user=phone;tag=314159
To: sip:+1-314-555-1111@ngw1.wcom.com
Call-Id: 12345600@ngw1.wcom.com
CSeq: 4 BYE
Content-Length: 0

REL F21
User C-> GW2

REL
CauseCode=16 Normal
CodingStandard=CCITT
RLC F22
GW2
->User C

RLC
6.1.2  Successful FGB PBX to ISDN PBX call with overflow

PBX User A calls PBX User C via GW1 using SS1 as a Proxy Server. During the attempt to reach User C via GW2, an error is encountered. SS1 receives a 503 Service Unavailable (F4) response to the forwarded INVITE. This could be due to all circuits being busy, or some other outage at GW2. SS1 recognizes the error and uses an alternative route via GW3 to terminate the call. From there, the call proceeds normally with User C answering the call. The call is terminated when User C hangs up.

Message Details

PBX A
->GW1

Seizure

GW1 -> PBX A

Wink
MF Digits F1
PBX A
->GW1

KP 444 3333 ST
INVITE F2
GW1 -> SS1

INVITE sip:444-3333,phone-context=p1234@ss1.wcom.com;user=phone
SIP/2.0
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com;user=phone
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Contact: PBX_A <sip:IdentifierString@gw1.wcom.com>
Content-Type: application/sdp
Content-Length: 136

v=0
c=IP4 2890844526 2890844526 IN IP4 gw1.wcom.com

c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* SS uses a location manager function to determine where B is actually located. Response is returned listing onnet and offnet routes. */

INVITE F3
SS1 -> GW2

INVITE sip:444-3333,phone-context=p1234@gw2.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:444-3333,phone-context=p1234@ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com;user=phone
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 136

v=0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

503 Service Unavailable F4
GW2 -> SS1

SIP/2.0 503 Service Unavailable
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=314159
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
ACK F5
SS1 -> GW2

ACK sip:444-3333,phone-context=p1234@gw2.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=314159
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

INVITE F6
SS1 -> GW3

INVITE sip:+1-918-555-3333@gw3.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:444-3333,phone-context=p1234@ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com;user=phone
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Contact: PBX_A <sip:IdentifierString@gw1.wcom.com>
Content-Type: application/sdp
Content-Length: 136

v=0
ci=PBX_A 2890844526 2890844526 IN IP4 gw1.wcom.com
t=0 0
c=IN IP4 100.101.102.103
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

SETUP F7
GW3-> PBX C

Protocol discriminator=Q.931
Call reference: Flag=0, CR value=any valid value not in use
Message type=SETUP
Bearer capability: Information transfer capability=0 (Speech) or 16
(3.1 kHz audio)
Channel identification=Preferred or exclusive B-channel
Progress indicator=1 (Call is not end-to-end ISDN; further call
progress information may be available inband)
Called party number:
Type of number and numbering plan ID=33 (National number in ISDN numbering plan)
Digits=918-555-3333

(100 Trying F8
GW3
->SS1)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com;user=phone
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

CALL PROCceeding F9
PBX C
-> GW3

Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F9 SETUP message
Message type=CALL PROC
Channel identification=Exclusive B-channel
PROGress F10
PBX C
-> GW3

Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F9 SETUP message
Message type=PROG
Progress indicator=1 (Call is not end-to-end ISDN; further call progress information may be available inband)

/* Based on PROGress message, GW3 returns a 183 response with SDP allowing in-band call progress indications to be sent to the originator. */

183 Session Progress F11
GW3 -> SS1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 134

v=0
c=PBX_B 987654321 987654321 IN IP4 gw3.wcom.com
t=0 0
c=IN IP4 100.101.102.104
m=audio 14918 RTP/AVP 0
a=rtpmap:0 PCMU/8000

183 Session Progress F12
SS1 -> GW1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 134

v=0
c=PBX_B 987654321 987654321 IN IP4 gw3.wcom.com
t=0 0
c=IN IP4 100.101.102.104
m=audio 14918 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* GW1 receives packets from GW3 with encoded ringback, tones or other audio. GW1 decodes this and places it on the originating trunk. */

CONNECT F13
PBX C
-> GW3

 Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F9 SETUP message
Message type=CONNECT
200 OK F14
GW3 -> SS1
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:444-3333,phone-context=p1234@ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-918-555-3333@gw3.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 134

v=0
o=PBX_B 987654321 987654321 IN IP4 gw3.wcom.com
t=0 0
c=IN IP4 100.101.102.104
m=audio 14918 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F15
SS1 -> GW1

SIP/2.0 200 OK
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:444-3333,phone-context=p1234@ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Contact: sip:+1-918-555-3333@gw3.wcom.com;user=phone
Content-Type: application/sdp
Content-Length: 134

v=0
o=PBX_B 987654321 987654321 IN IP4 gw3.wcom.com
t=0 0
c=IN IP4 100.101.102.104
m=audio 14918 RTP/AVP 0
a=rtpmap:0 PCMU/8000

GW1 -> PBX A

Seizure
ACK F16
GW1 -> SS1
ACK sip:+1-918-555-3333@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP gw1.wcom.com:5060
Route: <sip:+1-918-555-3333@gw3.wcom.com;user=phone>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:+1-918-555-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

ACK F17
SS1 -> GW3

ACK sip:+1-918-555-3333@gw3.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: sip:+1-918-555-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 ACK
Content-Length: 0

CONNect ACK F18
GW3-> PBX C

Protocol discriminator=Q.931
Call reference: Flag=0, CR value=value in F9 SETUP message
Message type=CONN ACK

/* RTP streams are established between GW1 and GW3. */

/ * User B Hangs Up with User A. */
DISConnect F19
PBX C
-> GW3

Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F9 SETUP message
Message type=DISC
Cause=16 (Normal clearing)
BYE F20
GW3 -> SS1

BYE sip:IdentifierString@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP gw3.wcom.com:5060
Route: <sip:IdentifierString@gw1.wcom.com>
From: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
To: PBX_A <sip:IdentifierString@gw1.wcom.com>
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 BYE
Content-Length: 0

BYE F21

SS1 -> GW1

BYE sip:IdentifierString@gw1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw3.wcom.com:5060
From: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
To: PBX_A <sip:IdentifierString@gw1.wcom.com>
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 BYE
Content-Length: 0

GW1 -> PBX A

Seizure removal
RELease F22
GW3-> PBX C

Protocol discriminator=Q.931
Call reference: Flag=0, CR value=value in F9 SETUP message
Message type=REL
200 OK F23

GW1 -> SS1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060
Via: SIP/2.0/UDP gw3.wcom.com:5060
From: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
To: PBX_A <sip:IdentifierString@gw1.wcom.com>
Call-Id: 12345600@gw1.wcom.com
CSeq: 1 BYE
Content-Length: 0
200 OK F24
SS11
-->GW3

SIP/2.0 200 OK
Via: SIP/2.0/UDP gw3.wcom.com:5060
From: sip:444-3333,phone-context=p1234@ss1.wcom.com
;user=phone;tag=123456789
To: PBX_A <sip:IdentifierString@gw1.wcom.com>
Call-ID: 12345600@gw1.wcom.com
CSeq: 1 BYE
Content-Length: 0

RELease COMplete F25
PBX C
-->GW3

Protocol discriminator=Q.931
Call reference: Flag=1, CR value=value in F9 SETUP message
Message type=REL COM
PBX A
-->GW1

Seizure removal
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References


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