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SIP Call Flow Examples

Status of this Memo

Abstract

This informational document gives examples of SIP (Session Initiation Protocol) call flows for IP telephony. 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). IP telephony 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 ISDN (Integrated Services Digital Network), ANSI ISUP (ISDN User Part), and FGB (Feature Group B) circuit associated signaling. PSTN calls are illustrated using global telephone numbers from the PSTN and private extensions served on by a PBX (Private Branch Exchange). Example SIP messages used for testing during SIP "bakeoff" events include SIP "torture test" messages, and messages with invalid parameters, methods, and tags.

Table of Contents

1 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. The message examples were developed during the SIP interoperability testing "bake-offs."

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.

Note that this document is informational, and is not normative on any aspect of SIP or SIP/PSTN interworking.

These call flows are based on the current version 2.0 of SIP in RFC2543[2].

Various PSTN signaling protocols are illustrated in this document: ISDN (Integrated Services Digital Network), ANSI ISUP (ISDN User Part) 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, some example messages are given along with expected behavior of clients and servers.

1.1 General Assumptions

A number of architecture, network, and protocol assumptions underlie the call flows in this document. Note that these assumptions are NOT requirements. They are outlined in this section so that they may be taken into consideration and to aide in the understanding of the call flow examples.

The authentication of SIP User Agents in these example call flows is performed using SIP Digest[2].

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[3] or some other non-SIP scheme.

The SIP Proxy Server has access to a Location Service 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. This allows them to implement features later in the call, which are not shown in these flows.

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. The actual names of Gateways will be vendor and implementation specific. However, two catagories are described here since the type of Gateway determines the form of the SIP URL used to identify them. The two types are:

. Network Gateway. This high port count PSTN gateway originates and terminates calls to the PSTN. Its 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@wcom.com) Alternatively, the "phone-context" qualifier could be used to interpret the private number. It is defined in the telephony URL document [4].
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. Note that the visual separator "-" is used purely to aid in the readability of the examples; a real gateway would be unlikely to insert visual separators.

. 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[4]. 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).

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

1.2 Legend for Message Flows

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

Double dashed lines (===) represent media paths between network elements.

Messages 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 RFC 2543[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 is used in SIP to Gateway and Gateway to Gateway via SIP calling (Sections 4 and 6). The 183 response indicates to the User Agent that a Gateway has been contacted and is trying to complete the call to the PSTN and that one-way early media may be present which gives an indication of the progress of the call. The User Agent will immediately play to the user any RTP media packets received to hear in-band call progress information such as ringtone or busy tone. Note that SDP is not required in the 183 response since the media is uni-directional. 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 content length calculations assume that each line of SDP ends with both a CR and a LF character.

. 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.

. Branch tags inserted by proxies are unique for each request. In this document, they are not due to editing simplications.

. Tags and Call-IDs are also reused for editing purposes.

. Other simplifications have been made for editing purposes. For example, the order of SIP headers is fairly consistent throughout the document. With a few exceptions, the ordering of SIP headers is not significant.

In any discrepency between this document and RFC 2543, follow RFC 2543. This document is informational only.

1.4 Changes to 00 draft

The major changes between this draft and the previous draft are listed below:

- SIP Telephony Service Examples have been removed from the draft. They will be revised using the TRANSFER header in a separate draft.

- Updated draft with RFC2543bis changes including: adding maddr to all Record-Route and Route headers, adding branch tags to Via headers inserted by proxies, added 487 response to CANCEL scenarios.

- Added example of INFO method in 5.1.1.

- Added Session: media to all 183 messages.

- Corrected a number of typos including putting user=phone tags inside <>, fixing Request-URI on PRACK, added missing tags, fixed Request-URIs that did not match To header in initial INVITE.

- Corrected all registrations to have same Call-ID.

1.5 Changes to 01 draft

The major changes between this draft and the previous draft are listed below:

- Added an example of multiple proxy authentication (3.1.3).

- Removed INFO method from 5.1.1. The use of INFO to transport DTMF is controversial.

- Fixed CSeq and Call-ID errors in Section 2.

- Replaced Session: media with Content-Disposition: session

- Added notes about CSeq numbering spaces.

- Fixed maddr in some Route headers.

- Corrected tag handling and Record-Routes in Section 4.

1.6 Changes to 02 draft

The major changes between this draft and previous are listed below:

- The word "Telephony" has been removed from the document title.

- Selected messages have been successfully validated against a SIP parser for accuracy.

- Added word "informational" to Abstract, and added additional wording about this document being informational and not normative.

- Added re-INVITE call flow (T.38 Fax)in 3.1.7.

- Removed SDP from all 183 Session Progress messages since it is not needed. Corrected figures to show early media path as one way.

- Removed PRACK from example 4.1.2 since there is no longer any need for the 183 to be transported reliably since it no longer contains SDP.

- Corrected Call-ID in 3.1.1.

- Corrected domain in WWW-Authenticate and Proxy-Authenticate headers to be a SIP URL.

- Section 7.4 SIP Date corrected to Sat, 01 Dec 2040 16:00:00 GMT

- Section 7.7 Retransmission strategy for unknown method message should be based on BYE not INVITE

- Section 7.10 multiple message element of this test removed as this behaviour is no longer allowed, higher SIP version number part moved into 7.42

- Section 7.18 recommended response changed

- Added Test Messages 7.21 through 7.41

1.7 Changes to the 03 draft

- Replaced Authorization headers with Proxy-Authorization headers

- Fixed Record-Route in example 3.1.3

- Various CSeq and Content-Length corrections

1.8 Changes to the 04 draft

- Removed domain names from Contact headers

- Removed "phone-context" URI parameter.

- Added expires parameter to 200 OK responses to REGISTERs.

- Changed Proxy-Authorization headers in Section 2 to Authorization headers.

2 SIP Registration Services

2.1 Success Scenarios

Registration either validates or invalidates a SIP client for user services provided by a SIP server. Additionally, the client provides one or more contact locations to the SIP server with the registration request. Registration is used by a Proxy to route incoming calls in an IP Telephony network. Registration are shown with authentication in these call flows. If authentication is not used, an imposter could "hijack" someone else's calls.

2.1.1 SIP Client New Registration

```
    User B                         SIP Server
      |                                |
      |          REGISTER F1           |
      |------------------------------->|
      |                                |
      |      401 Unauthorized F2       |
      |<-------------------------------|
      |                                |
      |          REGISTER F3           |
      |------------------------------->|
      |                                |
      |            200 OK F4           |
      |<-------------------------------|
      |                                |
```

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 RFC 2543[2]. It is assumed that User B has not previously registered with this Server.

Message Details

F1 REGISTER B -> SIP Server

```
REGISTER sip:ss2.wcom.com SIP/2.0
```

```
   Via: SIP/2.0/UDP there.com:5060
   From: LittleGuy <sip:UserB@there.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 123456789@there.com
   CSeq: 1 REGISTER
   Contact: <sip:UserB@110.111.112.113>
   Contact: <sip:+1-972-555-2222@gw1.wcom.com;user=phone>
   Contact: tel:+1-972-555-2222
   Content-Length: 0


   F2 401 Unauthorized SIP Server -> User B

   SIP/2.0 401 Unauthorized
   Via: SIP/2.0/UDP there.com:5060
   From: LittleGuy <sip:UserB@there.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 123456789@there.com
   CSeq: 1 REGISTER
   WWW-Authenticate: Digest realm="MCI WorldCom SIP",
    domain="sip:ss2.wcom.com", nonce="ea9c8e88df84f1cec4341ae6cbe5a359",
    opaque="", stale=FALSE, algorithm=MD5
   Content-Length: 0


   F3 REGISTER B -> SIP Server

   REGISTER sip:ss2.wcom.com SIP/2.0
   Via: SIP/2.0/UDP there.com:5060
   From: LittleGuy <sip:UserB@there.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 123456789@there.com
   CSeq: 2 REGISTER
   Contact: <sip:UserB@110.111.112.113>
   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


   F4 200 OK SIP Server -> B

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP there.com:5060
   From: LittleGuy <sip:UserB@there.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 123456789@there.com
   CSeq: 2 REGISTER
```

```
Contact: <sip:UserB@110.111.112.113>;expires=3600
Contact: <sip:+1-972-555-2222@gw1.wcom.com;user=phone>;expires=3600
Contact: <tel:+1-972-555-2222>;expires=4294967295
Content-Length: 0
```

2.1.2 User updates contact list

```
User B                        SIP Server
  |                                |
  |          REGISTER F1           |
  |------------------------------->|
  |                                |
  |            200 OK F2           |
  |<-------------------------------|
  |                                |
```

User B wishes to update the list of addresses where the SIP server will redirect or forward INVITE requests.

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

F1 REGISTER B -> SIP Server

```
REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
CSeq: 1 REGISTER
Contact: mailto:UserB@there.com
Authorization:Digest username="UserB",
 realm="MCI WorldCom SIP",
 nonce="1cec4341ae6cbe5a359ea9c8e88df84f", opaque="",
 uri="sip:ss2.wcom.com", response="71ba27c64bd01de719686aa4590d5824"
Content-Length: 0
```

F2 200 OK SIP Server -> B

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
```

```
   CSeq: 1 REGISTER
   Contact: <sip:UserB@110.111.112.113>;expires=3600
   Contact: <sip:+1-972-555-2222@gw1.wcom.com;user=phone>;expires=3600
   Contact: <tel:+1-972-555-2222>;expires=4294967295
   Contact: <mailto:UserB@there.com>;expires=4294967295
   Content-Length: 0
```

2.1.3 User Requests Current Contact List

```
User B                        SIP Server
  |                               |
  |          REGISTER F1          |
  |------------------------------>|
  |                               |
  |            200 OK F2          |
  |<------------------------------|
  |                               |
```

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. The server returns a response (200 OK) which includes the user’s current registration list in Contact headers.

Message Details

F1 REGISTER B -> SIP Server

```
REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
CSeq: 1 REGISTER
Authorization:Digest username="UserB",
 realm="MCI WorldCom SIP",
 nonce="df84f1cec4341ae6cbe5ap359a9c8e88", opaque="",
 uri="sip:ss2.wcom.com", response="aa7ab4678258377c6f7d4be6087e2f60"
Content-Length: 0
```

F2 200 OK SIP Server -> B

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
CSeq: 1 REGISTER
Contact: <sip:UserB@110.111.112.113>;expires=3600
Contact: <sip:+1-972-555-2222@gw1.wcom.com;user=phone>;expires=3600
Contact: <tel:+1-972-555-2222>;expires=4294967295
Contact: <mailto:UserB@there.com>;expires=4294967295
```

```
Content-Length: 0
```

2.1.4 User Cancels Registration

```
User B                         SIP Server
  |                                |
  |          REGISTER F1           |
  |------------------------------->|
  |                                |
  |            200 OK F2           |
  |<-------------------------------|
  |                                |
```

User B wishes to cancel their registration with the SIP 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

F1 REGISTER B -> SIP Server

```
REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
CSeq: 1 REGISTER
Expires: 0
Contact: *
Authorization:Digest username="UserB", realm="MCI WorldCom SIP",
 nonce="88df84f1cac4341aea9c8ee6cbe5a359", opaque="",
 uri="sip:ss2.wcom.com", response="ff0437c51696f9a76244f0cf1dbabbea"
Content-Length: 0
```

F2 200 OK SIP Server -> B

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
CSeq: 1 REGISTER
Content-Length: 0
```

2.2 Failure Scenarios

2.2.1 Unsuccessful SIP registration

```
   User B                        SIP Server
     |                                |
     |          REGISTER F1           |
     |------------------------------->|
     |                                |
     |      401 Unauthorized F2       |
     |<-------------------------------|
     |                                |
     |          REGISTER F3           |
     |------------------------------->|
     |                                |
     |      401 Unauthorized F4       |
     |<-------------------------------|
     |                                |
```

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

F1 REGISTER B -> SIP Server

```
REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
CSeq: 1 REGISTER
Contact: <sip:UserB@110.111.112.113>
Contact: <sip:+1-972-555-2222@gw1.wcom.com;user=phone>
Contact: tel:+1-972-555-2222
Content-Length: 0
```

F2 Unauthorized SIP Server -> User B

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
CSeq: 1 REGISTER
WWW-Authenticate: Digest realm="MCI WorldCom SIP",
 domain="sip:ss2.wcom.com", once="f1cec4341ae6ca9c8e88df84be55a359",
 opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0


F3 REGISTER B -> SIP Server

REGISTER sip:ss2.wcom.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
CSeq: 2 REGISTER
Contact: <sip:UserB@110.111.112.113>
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="f1cec4341ae6ca9c8e88df84be55a359", opaque="",
 uri="sip:ss2.wcom.com", response="61f8470ceb87d7ebf508220214ed438b"
Content-Length: 0


/*  The response above encodes the incorrect password */

F4 401 Unauthorized SIP Server -> User B

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456789@there.com
CSeq: 2 REGISTER
WWW-Authenticate: Digest realm="MCI WorldCom SIP",
 domain="sip:ss2.wcom.com", nonce="84f1c1ae6cbe5ua9c8e88dfa3ecm3459",
 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 Agents (UAs): User A and User B. User A (LittleGuy sip:UserA@here.com) and User B (BigGuy sip:UserB@there.com) are assumed to be SIP phones or SIP-enabled devices. 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 Proxy Servers to authenticate the caller User A. It is assumed that User B has registered with Proxy Server Proxy 2 as per Section 2.1 to be able to receive the calls via the Proxy.

3.1.1 Successful Simple SIP to SIP

```
   User A                  User B
     |                        |
     |       INVITE F1        |
     |----------------------->|
     |    (100 Trying) F2     |
     |<-----------------------|
     |    180 Ringing F3      |
     |<-----------------------|
     |                        |
     |       200 OK F4        |
     |<-----------------------|
     |         ACK F5         |
     |----------------------->|
     |   Both Way RTP Media   |
     |<======================>|
     |                        |
     |         BYE F6         |
     |<-----------------------|
     |       200 OK F7        |
     |----------------------->|
     |                        |
```

In this scenario, User A completes a call to User B directly.

Message Details

F1 INVITE User A -> User B

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 (100 Trying) User B -> User A

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F3 180 Ringing User B -> User A

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=8321234356
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F4 200 OK User B -> User A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=8321234356
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 147

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

F5 ACK User A -> User B

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=8321234356
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. Note that the CSeq is NOT 2, since
   User A and User B maintain their own independent CSeq counts.
   (The INVITE was request 1 generated by User A, and the BYE is
   request 1 generated by User B) */

F6 BYE User B -> User A

BYE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=8321234356
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0


F7 200 OK User A -> User B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=8321234356
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0
```

3.1.2 Successful SIP to SIP through two proxies

```
   User A          Proxy 1          Proxy 2          User B
     |                |                |                |
     |   INVITE F1    |                |                |
     |--------------->|                |                |
     |     407 F2     |                |                |
     |<---------------|                |                |
     |     ACK F3     |                |                |
     |--------------->|                |                |
     |   INVITE F4    |                |                |
     |--------------->|   INVITE F5    |                |
     |    (100) F6    |--------------->|   INVITE F7    |
     |<---------------|    (100) F8    |--------------->|
     |                |<---------------|                |
     |                |                |     180 F9     |
     |                |    180 F10     |<---------------|
     |     180 F11    |<---------------|                |
     |<---------------|                |     200 F12    |
     |                |    200 F13     |<---------------|
     |     200 F14    |<---------------|                |
     |<---------------|                |                |
     |     ACK F15    |                |                |
     |--------------->|    ACK F16     |                |
     |                |--------------->|     ACK F17    |
     |                |                |--------------->|
     |                Both Way RTP Media                |
     |<================================================>|
     |                |                |     BYE F18    |
     |                |    BYE F19     |<---------------|
     |     BYE F20    |<---------------|                |
     |<---------------|                |                |
     |     200 F21    |                |                |
     |--------------->|     200 F22    |                |
     |                |--------------->|     200 F23    |
     |                |                |--------------->|
     |                |                |                |
```

In this scenario, User A completes a call to User B using two proxies Proxy 1 and Proxy 2. The initial INVITE (F1) does not contain the Authorization credentials Proxy 1 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.

Proxy 1 inserts a Record-Route header into the INVITE message to ensure that it is present in all subsequent message exchanges. Proxy 2 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

F1 INVITE A -> Proxy 1

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 challenges User A for authentication */

F2 407 Proxy Authorization Required Proxy 1 -> User A

```
SIP/2.0 407 Proxy Authorization Required
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Proxy-Authenticate: Digest realm="MCI WorldCom SIP",
 domain="sip:ss1.wcom.com", nonce="f84f1cec41e6cbe5aea9c8e88d359",
 opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0
```

F3 ACK A -> Proxy 1

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
```

```
   CSeq: 1 ACK
   Content-Length: 0


   /* User A responds be re-sending the INVITE with authentication
      credentials in it.  A new Call-ID is used, so the CSeq is reset
      back to 1 */

   F4 INVITE A -> Proxy 1

   INVITE sip:UserB@there.com SIP/2.0
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 12345601@here.com
   CSeq: 1 INVITE
   Contact: <sip:UserA@100.101.102.103>
   Authorization:Digest username="UserA",
    realm="MCI WorldCom SIP",
    nonce="wf84f1ceczx41ae6cbe5aea9c8e88d359", opaque="",
    uri="sip:ss1.wcom.com", response="42ce3cef44b22f50c6a6071bc8"
   Content-Type: application/sdp
   Content-Length: 147

   v=0
   o=UserA 2890844526 2890844526 IN IP4 here.com
   s=Session SDP
   c=IN IP4 100.101.102.103
   t=0 0
   m=audio 49172 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 49172 from the
   network. */

   F5 INVITE Proxy 1 -> Proxy 2

   INVITE sip:UserB@there.com SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP here.com:5060
   Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 12345601@here.com
   CSeq: 1 INVITE
   Contact: <sip:UserA@100.101.102.103>
   Content-Type: application/sdp
   Content-Length: 147
```

```
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F6 (100 Trying) Proxy 1 -> User A

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F7 INVITE Proxy 2 -> B

```
INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F8 (100 Trying) Proxy 2 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
```

```
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F9 180 Ringing B -> Proxy 2

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F10 180 Ringing Proxy 2 -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F11 180 Ringing Proxy 1 -> A

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

F12 200 OK B -> Proxy 2

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
```

```
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 147

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

F13 200 OK Proxy 2 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 147

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

F14 200 OK Proxy 1 -> A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 1 INVITE
```

```
   Contact: <sip:UserB@110.111.112.113>
   Content-Type: application/sdp
   Content-Length: 147

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


   F15 ACK A -> Proxy 1

   ACK sip:UserB@there.com SIP/2.0
   Via: SIP/2.0/UDP here.com:5060
   Route: <sip:UserB@there.com;maddr=ss2.wcom.com>,
    <sip:UserB@110.111.112.113>
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=314159
   Call-ID: 12345601@here.com
   CSeq: 1 ACK
   Content-Length: 0


   F16 ACK Proxy 1 -> Proxy 2

   ACK sip:UserB@there.com SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP here.com:5060
   Route: <sip:UserB@110.111.112.113>
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=314159
   Call-ID: 12345601@here.com
   CSeq: 1 ACK
   Content-Length: 0


   F17 ACK Proxy 2 -> B

   ACK sip:UserB@110.111.112.113 SIP/2.0
   Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <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. */

/* Again, note that the CSeq is NOT 2. User A and User B maintain their own separate CSeq counts */

F18 BYE User B -> Proxy 2

```
BYE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com;maddr=ss1.wcom.com>,
 <sip:UserA@100.101.102.103>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0
```

F19 BYE Proxy 2 -> Proxy 1

```
BYE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@100.101.102.103>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0
```

F20 BYE Proxy 1 -> User A

```
BYE sip:UserA@100.101.102.103 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0
```

F21 200 OK User A -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
```

```
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0
```

F22 200 OK Proxy 1 -> Proxy 2

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0
```

F23 200 OK Proxy 2 -> User B

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 BYE
Content-Length: 0
```

3.1.3 SIP to SIP with Multi Proxy Authentication

```
     User A       Proxy 1     Proxy 2       User B
       |  INVITE F1 |           |             |
       |----------->|           |             |
       |  407 Proxy Authorization Required F2 |
       |<-----------|           |             |
       |   ACK F3   |           |             |
       |----------->|           |             |
       |  INVITE F4 |           |             |
       |----------->|           |             |
       |  (100 F5)  |           |             |
       |<-----------| INVITE F6 |             |
       |            |---------->|             |
       |            |  407 Proxy Authorization Required F7
       |            |<----------|             |
       |            |   ACK F8  |             |
       |            |---------->|             |
       |  407 Proxy Authorization Required F9 |
       |<-----------|           |             |
       |   ACK F10  |           |             |
       |----------->|           |             |
       |  INVITE F11|           |             |
       |----------->|           |             |
       |  (100 F12) |           |             |
       |<-----------| INVITE F13|             |
       |            |---------->|             |
       |            | (100 F14) |             |
       |            |<----------|  INVITE F15 |
       |            |           |------------>|
       |            |           | 200 OK F16  |
       |            | 200 OK F17|<------------|
       | 200 OK F18 |<----------|             |
       |<-----------|           |             |
       |   ACK F19  |           |             |
       |----------->|  ACK F20  |             |
       |            |---------->|   ACK F21   |
       |            |           |------------>|
       |           RTP Media Path             |
       |<====================================>|
```

In this scenario, User A completes a call to User B using two proxies Proxy 1 and Proxy 2. The initial INVITE (F1) does not contain the Authorization credentials Proxy 1 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 after Proxy 2 challenges and receives valid credentials. The call terminates when User B disconnects by initiating a BYE message.

Proxy 1 inserts a Record-Route header into the INVITE message to ensure that it is present in all subsequent message exchanges. Proxy 2 also inserts itself into the Record-Route header.

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 challenges User A for authentication */

F2 407 Proxy Authorization Required Proxy 1 -> User A

```
SIP/2.0 407 Proxy Authorization Required
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Proxy-Authenticate: Digest realm="WorldCom SIP",
 domain="sip:ss1.wcom.com", nonce="wf84f1cczx41ae6cbeaea9ce88d359",
 opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0
```

F3 ACK A -> Proxy 1

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
```

```
   Call-ID: 12345600@here.com
   CSeq: 1 ACK
   Content-Length: 0


   /* User A responds be re-sending the INVITE with authentication
      credentials in it.  The same Call-ID is used, so the CSeq is
      increased. */

   F4 INVITE A -> Proxy 1

   INVITE sip:UserB@there.com SIP/2.0
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 12345600@here.com
   CSeq: 2 INVITE
   Contact: <sip:UserA@100.101.102.103>
   Proxy-Authorization:Digest username="UserA", realm="WorldCom SIP",
    nonce="wf84f1ceczx41ae6cbe5aea9c8e88d359", opaque="",
    uri="sip:ss1.wcom.com", response="42ce3cef44b22f50c6a6071bc8"
   Content-Type: application/sdp
   Content-Length: 147

   v=0
   o=UserA 2890844526 2890844526 IN IP4 here.com
   s=Session SDP
   c=IN IP4 100.101.102.103
   t=0 0
   m=audio 49172 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 49172 from the
   network. */

   F5 (100 Trying Proxy 1 -> User A)

   SIP/2.0 100 Trying
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 12345600@here.com
   CSeq: 2 INVITE
   Content-Length: 0


   F6 INVITE Proxy 1 -> Proxy 2
```

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=230f2.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000


/* Proxy 2 challenges User A for authentication */

F7 407 Proxy Authorization Required Proxy 2 -> Proxy 1

SIP/2.0 407 Proxy Authorization Required
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=230f2.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=838209
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Proxy-Authenticate: Digest realm="MCI SIP",
 domain="sip:ss2.mci.com", nonce="c1e22c41ae6cbe5ae983a9c8e88d359",
 opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0


F8 ACK Proxy 1 -> Proxy 2

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=838209
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0
```

```
/* Proxy 1 forwards the challenge to User A for authentication from
Proxy 2 */

F9 407 Proxy Authorization Required Proxy 1 -> User A

SIP/2.0 407 Proxy Authorization Required
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=2341d
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Proxy-Authenticate: Digest realm="MCI SIP",
 domain="sip:ss2.mci.com", nonce="c1e22c41ae6cbe5ae983a9c8e88d359",
 opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0


F10 ACK User A -> Proxy 1

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=2341d
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0


/* User A responds be re-sending the INVITE with authentication
credentials for Proxy 1 AND Proxy 2.  */

F11 INVITE A -> Proxy 1

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 3 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA", realm="WorldCom SIP",
 nonce="wf84f1ceczx41ae6cbe5aea9c8e88d359", opaque="",
 uri="sip:ss1.wcom.com", response="42ce3cef44b22f50c6a6071bc8"
Proxy-Authorization:Digest username="UserA", realm="MCI SIP",
 nonce="c1e22c41ae6cbe5ae983a9c8e88d359", opaque="",
 uri="sip:ss2.mci.com", response="f44ab22f150c6a56071bce8"
Content-Type: application/sdp
Content-Length: 147

v=0
```

```
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 finds its credentials and authorizes User A, forwarding the INVITE to Proxy. */

F12 (100 Trying Proxy 1 -> User A)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 3 INVITE
Content-Length: 0
```

F13 INVITE Proxy 1 -> Proxy 2

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=230f2.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 3 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA", realm="MCI SIP",
 nonce="c1e22c41ae6cbe5ae983a9c8e88d359", opaque="",
 uri="sip:ss2.mci.com", response="f44ab22f150c6a56071bce8"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 2 finds its credentials and authorizes User A, forwarding the INVITE to User B. */

F14 (100 Trying Proxy 2 -> Proxy 1)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=230f2.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 3 INVITE
Content-Length: 0
```

F15 INVITE Proxy 2 -> User B

```
INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.mci.com:5060;branch=31972.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=230f2.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.mci.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 3 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* User B answers the call immediately */

F16 200 OK User B -> Proxy 2

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.mci.com:5060;branch=31972.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=230f2.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.mci.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=9103874
Call-ID: 12345600@here.com
```

```
CSeq: 3 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 149

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

F17 200 OK Proxy 2 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=230f2.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.mci.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=9103874
Call-ID: 12345600@here.com
CSeq: 3 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 149

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

F18 200 OK Proxy 1 -> User A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.mci.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=9103874
Call-ID: 12345600@here.com
CSeq: 3 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 149
```

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

F19 ACK User A -> Proxy 1

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@there.com;maddr=ss2.mci.com>,
 <sip:UserB@there.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=9103874
Call-ID: 12345600@here.com
CSeq: 3 ACK
Content-Length: 0
```

F20 ACK Proxy 1 -> Proxy 2

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=230f2.1
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@110.111.112.113>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=9103874
Call-ID: 12345600@here.com
CSeq: 3 ACK
Contact: <sip:UserB@110.111.112.113>
Content-Length: 0
```

F21 ACK Proxy 2 -> User A

```
ACK sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.mci.com:5060;branch=31972.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=230f2.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=9103874
Call-ID: 12345600@here.com
CSeq: 3 ACK
Contact: <sip:UserB@110.111.112.113>
Content-Length: 0
```

3.1.4 Successful SIP to SIP with Proxy failure

```
   User A          Proxy 1          Proxy 2          User B
     |                |                |                |
     |   INVITE F1    |                |                |
     |--------------->|                |                |
     |   INVITE F2    |                |                |
     |--------------->|                |                |
     |   INVITE F3    |                |                |
     |--------------->|                |                |
     |   INVITE F4    |                |                |
     |--------------->|                |                |
     |   INVITE F5    |                |                |
     |--------------->|                |                |
     |   INVITE F6    |                |                |
     |--------------->|                |                |
     |   INVITE F7    |                |                |
     |--------------->|                |                |
     |    CANCEL F8   |                |                |
     |--------------->|                |                |
     |     INVITE F9                   |                |
     |-------------------------------->|                |
     |            407 F10              |                |
     |<--------------------------------|                |
     |             ACK F11             |                |
     |-------------------------------->|                |
     |     INVITE F12                  |                |
     |-------------------------------->|   INVITE F13   |
     |            (100) F14            |--------------->|
     |<--------------------------------|                |
     |                                 |     180 F15    |
     |             180 F16             |<---------------|
     |<--------------------------------|                |
     |                                 |     200 F17    |
     |             200 F18             |<---------------|
     |<--------------------------------|                |
     |             ACK F19             |                |
     |-------------------------------->|     ACK F20    |
     |                                 |--------------->|
     |                Both Way RTP Media                |
     |<================================================>|
     |                                 |     BYE F21    |
     |             BYE F22             |<---------------|
     |<--------------------------------|                |
     |             200 F23             |                |
     |-------------------------------->|     200 F24    |
     |                                 |--------------->|
     |                                 |                |
```

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 Proxy 1 and a secondary SIP Proxy Server Proxy 2 (Or is able to use DNS SRV records to locate Proxy 1 and Proxy 2). Proxy 1 is out of service and does not respond to INVITEs (it is reachable, but unresponsive). After sending a CANCEL to Proxy 1, User A then completes the call to User B using Proxy 2.

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 INVITE A -> Proxy 1

Same as Message F1

F3 INVITE A -> Proxy 1

Same as Message F1

F4 INVITE A -> Proxy 1

Same as Message F1

F5 INVITE A -> Proxy 1

Same as Message F1

F6 INVITE A -> Proxy 1

Same as Message F1

F7 INVITE A -> Proxy 1

Same as Message F1

/* User A gives up on the unresponsive proxy and sends a CANCEL. */

F8 CANCEL A -> Proxy 1

```
CANCEL sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
```

F9 INVITE A -> Proxy 2

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 2 challenges User A for authentication */

F10 407 Proxy Authorization Required Proxy 2 -> User A

```
SIP/2.0 407 Proxy Authorization Required
Via: SIP/2.0/UDP here.com:5060
```

```
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Proxy-Authenticate: Digest realm="MCI SIP",
 domain="sip:ss2.wcom.com", nonce="1ae6cbe5ea9c8e8df84fqnlec434a359",
 opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0
```

F11 ACK A -> Proxy 2

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 ACK
Content-Length: 0
```

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

F12 INVITE A -> Proxy 2

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345602@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA", realm="MCI SIP",
 nonce="1ae6cbe5ea9c8e8df84fqnlec434a359", opaque="",
 uri="sip:ss2.wcom.com", response="8a880c919d1a52f20a1593e228adf599"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 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 49172 from the network. */

F13 INVITE Proxy 2 -> B

```
INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345602@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F14 (100 Trying) Proxy 2 -> User A

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345602@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F15 180 Ringing B -> Proxy 2

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345602@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F16 180 Ringing Proxy 2 -> A

```
SIP/2.0 180 Ringing
```

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

F17 200 OK B -> Proxy 2

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345602@here.com
CSeq: 1 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 149

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

F18 200 OK Proxy 2 -> A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345602@here.com
CSeq: 1 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 149

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

F19 ACK A -> Proxy 2

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@110.111.112.113>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345602@here.com
CSeq: 1 ACK
Content-Length: 0
```

F20 ACK Proxy 2 -> B

```
ACK sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <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. */

F21 BYE User B -> Proxy 2

```
BYE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@100.101.102.103>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345602@here.com
CSeq: 1 BYE
Content-Length: 0
```

F22 BYE Proxy 2 -> User A

```
BYE sip:UserA@100.101.102.103 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345602@here.com
CSeq: 1 BYE
```

```
Content-Length: 0


F23 200 OK User A -> Proxy 2

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345602@here.com
CSeq: 1 BYE
Content-Length: 0


F24 200 OK Proxy 2 -> User B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345602@here.com
CSeq: 1 BYE
Content-Length: 0
```

3.1.5 Successful SIP to SIP through SIP Firewall Proxy

```
User A          FW Proxy         Proxy 1          User B
  |                |                |                |
  |   INVITE F1    |                |                |
  |--------------->|   INVITE F2    |                |
  |    (100) F3    |--------------->|   INVITE F4    |
  |<---------------|    (100) F5    |--------------->|
  |                |<---------------|      180 F6    |
  |                |     180 F7     |<---------------|
  |     180 F8     |<---------------|                |
  |<---------------|                |      200 F9    |
  |                |    200 F10     |<---------------|
  |     200 F11    |<---------------|                |
  |<---------------|                |                |
  |     ACK F12    |                |                |
  |--------------->|     ACK F13    |                |
  |                |--------------->|     ACK F14    |
  |                |                |--------------->|
  |    RTP Media   |        Both Way RTP Media       |
  |<==============>|<===============================>|
  |     BYE F15    |                |                |
  |--------------->|     BYE F16    |                |
  |                |--------------->|     BYE F17    |
  |                |                |--------------->|
  |                |                |     200 F18    |
  |                |     200 F19    |<---------------|
  |     200 F20    |<---------------|                |
  |<---------------|                |                |
  |                |                |                |
```

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 (F10) messages, and possibly any 18x or ACK messages containing SDP.

In addition to firewall traversal, this Back-to-Back User Agent (B2BUA) 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

F1 INVITE A -> SIP FW

```
INVITE sip:UserB@there.com;maddr=fwp1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
```

```
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
 nonce="85b4f1cen4341ae6cbe5a3a9c8e88df9", opaque="",
 uri="sip:ss1.wcom.com", response="b3f392f9218a328b9294076d708e6815"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Client for A prepares to receive data on port 49172 from the network. */

F2 INVITE SIP FW -> Proxy 1

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=fwp1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
 nonce="85b4f1cen4341ae6cbe5a3a9c8e88df9", opaque="",
 uri="sip:ss1.wcom.com", response="b3f392f9218a328b9294076d708e6815"
Content-Type: application/sdp
Content-Length: 149

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

```
F3 (100 Trying) SIP FW -> A

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


/* SIP FW prepares to proxy data from port 200.201.202.203/1000 to
100.101.102.103/49172.   Proxy 1 uses a Location Service function to
determine where B is located. Based upon location analysis the call
is forwarded to User B */

F4 INVITE Proxy 1 -> B

INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>,
 <sip:UserB@there.com;maddr=fwp1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 146

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


F5 (100 Trying) Proxy 1 -> SIP FW

SIP/2.0 100 Trying
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 180 Ringing B -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F7 180 Ringing Proxy 1 -> SIP FW

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F8 180 Ringing SIP FW -> A

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

F9 200 OK B -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>,
 <sip:UserB@there.com;maddr=fwp1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserB@110.111.112.113>
```

```
Content-Type: application/sdp
Content-Length: 147

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


F10 200 OK Proxy 1 -> SIP FW

SIP/2.0 200 OK
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>,
 <sip:UserB@there.com;maddr=fwp1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 147

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


F11 200 OK SIP FW -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>,
 <sip:UserB@there.com;maddr=fwp1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 147

v=0
```

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


   /* The Firewall Proxy prepares to proxy packets from 200.201.202.203/
      1002 to 110.111.112.113/3456 */

   F12 ACK A -> SIP FW

   ACK sip:UserB@there.com;maddr=fwp1.wcom.com SIP/2.0
   Via: SIP/2.0/UDP here.com:5060
   Route: <sip:UserB@there.com;maddr=ss1.wcom.com>,
    <sip:UserB@110.111.112.113>
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 1 ACK
   Content-Length: 0


   F13 ACK SIP FW -> Proxy 1

   ACK sip:UserB@there.com SIP/2.0
   Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
   Via: SIP/2.0/UDP here.com:5060
   Route: <sip:UserB@110.111.112.113>
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 1 ACK
   Content-Length: 0


   F14 ACK Proxy 1 -> B

   ACK sip:UserB@110.111.112.113 SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <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 Firewall Proxy and
```

```
between the Firewall Proxy and B*/

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

F15 BYE A -> SIP FW

BYE sip:UserB@fwp1.wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@there.com;maddr=ss1.wcom.com>,
 <sip:UserB@110.111.112.113>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0


F16 BYE SIP FW -> Proxy 1

BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@110.111.112.113>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0


F17 BYE F18 Proxy 1 -> B

BYE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0


F18 200 OK B -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
```

```
   Call-ID: 12345600@here.com
   CSeq: 2 BYE
   Content-Length: 0
```

F19 200 OK Proxy 1 -> SIP FW

```
   SIP/2.0 200 OK
   Via: SIP/2.0/UDP fwp1.wcom.com:5060;branch=9471385739578.1
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 2 BYE
   Content-Length: 0
```

F20 200 OK SIP FW -> A

```
   SIP/2.0 200 OK
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 2 BYE
   Content-Length: 0
```

3.1.6 Successful SIP to SIP via Redirect and Proxy with SDP in ACK

```
   User A       Redirect Proxy      Proxy 2           User B
     |                |                |                |
     |   INVITE F1    |                |                |
     |--------------->|                |                |
     |     302 F2     |                |                |
     |<---------------|                |                |
     |     ACK F3     |                |                |
     |--------------->|                |                |
     |     INVITE F4                   |                |
     |-------------------------------->|    INVITE F5   |
     |            (100) F6             |--------------->|
     |<--------------------------------|    (100) F7    |
     |                                 |<---------------|
     |                                 |     180 F8     |
     |             180 F9              |<---------------|
     |<--------------------------------|                |
     |                                 |    200 F10     |
     |             200 F11             |<---------------|
     |<--------------------------------|                |
     |             ACK F12             |                |
     |-------------------------------->|    ACK F13     |
     |                                 |--------------->|
     |                Both Way RTP Media                |
     |<================================================>|
     |                                 |    BYE F14     |
     |             BYE F15             |<---------------|
     |<--------------------------------|                |
     |             200 F16             |                |
     |-------------------------------->|    200 F17     |
     |                                 |--------------->|
     |                                 |                |
```

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. In this example, no SDP is present in the INVITE, so the SDP is carried in the ACK message.

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

Message Details

F1 INVITE A -> Redirect Proxy

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Length: 0
```

F2 302 Moved Temporarily Redirect Proxy -> A

```
SIP/2.0 302 Moved Temporarily
Contact: sip:UserB@everywhere.com
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F3 ACK A -> Redirect Proxy

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

F4 INVITE A -> Proxy 2

```
INVITE sip:UserB@everywhere.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Length: 0
```

F5 INVITE Proxy 2 -> B

```
INVITE sip:UserB@111.112.113.114 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@everywhere.com;maddr=ss2.wcom.com>
```

```
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Length: 0
```

F6 (100 Trying) Proxy 2 -> A

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0
```

F7 (100 Trying) B -> Proxy 2

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0
```

F8 180 Ringing B -> Proxy 2

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0
```

F9 180 Ringing Proxy 2 -> A

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 INVITE
```

```
Content-Length: 0
```

F10 200 OK B -> Proxy 2

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@everywhere.com;maddr=ss2.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: <sip:UserB@111.112.113.114>
Content-Type: application/sdp
Content-Length: 152

v=0
o=UserB 2890844527 2890844527 IN IP4 everywhere.com
s=Session SDP
c=IN IP4 111.112.113.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F11 200 OK Proxy -> A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: <sip:UserB@111.112.113.114>
Content-Type: application/sdp
Content-Length: 152

v=0
o=UserB 2890844527 2890844527 IN IP4 everywhere.com
s=Session SDP
c=IN IP4 111.112.113.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* ACK contains SDP of A since none present in INVITE */

F12 ACK A -> Proxy 2

```
ACK sip:UserB@everyhere.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@111.112.113.114>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 ACK
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F13 ACK Proxy 2 -> B

```
ACK sip:UserB@111.112.112.114 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 ACK
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

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

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

F14 BYE B -> Proxy 2

```
BYE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP everywhere.com:5060
Route: <sip:UserA@100.101.102.103>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345600@here.com
```

```
CSeq: 1 BYE
Content-Length: 0
```

F15 BYE Proxy 2 -> A

```
BYE sip:UserA@100.101.102.103 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP everywhere.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0
```

F16 200 OK A -> Proxy 2

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP everywhere.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0
```

F17 200 OK Proxy 2 -> B

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP everywhere.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0
```

3.1.7 Successful SIP to SIP with re-INVITE (T.38 Fax)

```
       IFT UA                 Proxy              IFTGW UA
           |   F1 INVITE        |                    |
           |------------------->|      F2 INVITE     |
           |                    |------------------->|
           |   F3  100 Trying   |                    |
           |<-------------------|   F4  100 Trying   |
           |                    |<-------------------|
           |                    |   F5  180 Ringing  |
           |   F6  180 Ringing  |<-------------------|
           |<-------------------|                    |
           |                    |   F7  200 OK       |
           |   F8  200 OK       |<-------------------|
           |<-------------------|                    |
           |   F9  ACK          |                    |
           |------------------->|   F10 ACK          |
           |                     ------------------->|
           |      Both Way RTP Media Established     |
           |<=======================================>|
           |                    |   F11 INVITE       |
           |    F12 INVITE      |<-------------------|
           |<-------------------|                    |
           |   F13 200 OK       |                    |
           |------------------->|    F14 200 OK      |
           |                    |------------------->|
           |                    |   F15  ACK         |
           |      F16  ACK      |<-------------------|
           |<-------------------|                    |
           |          T.38/UDPT Fax Flow Established |
           |<=======================================>|
           |                    |   F17 INVITE       |
           |     F18 INVITE     |<-------------------|
           |<-------------------|                    |
           |   F19 200 OK       |                    |
           |------------------->|     F20 200 OK     |
           |                    |------------------->|
           |                    |   F21  ACK         |
           |           F22  ACK |<-------------------|
           |<-------------------|                    |
           |          RTP Media Re-Established       |
           |<=======================================>|
           |   F23 BYE          |                    |
           |------------------->|       F24 BYE      |
           |                    |------------------->|
           |                    |   F25  200 OK      |
           |    F26  200 OK     |<-------------------|
           |<-------------------|                    |
```

This example shows a session whose media session changes twice during the session. The initial PCM media session is established between the two User Agents in messages F1 through F10. The called party then wishes to change the type of media session between them. (In this example, the called party detects a fax transmission and wishes to change the media session to a T.38 Fax over IP session). The called party then sends a re-INVITE in message F11 which is accepted in the 200 OK of message F13. After message F16, the original PCM media session is terminated and the new T.38 session established. After the fax transmission is over, the called party again wishes to switch the media session back to a PCM media session and initiates a re-INVITE sequence with message F17.

Message Details

F1 INVITE IFT UA -> PROXY

```
INVITE sip:+1-650-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com;user=phone>
Content-Type: application/sdp
Content-Length: 146

v=0
o=IFAXTERMINAL01 2890844527 2890844527 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 INVITE PROXY -> IFTGW UA

```
INVITE sip:+1-650-555-2222@iftgw.there.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com;
    maddr=ss1.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com;user=phone>
Content-Type: application/sdp
Content-Length: 146
```

```
v=0
o=IFAXTERMINAL01 2890844527 2890844527 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F3 (100 Trying) PROXY -> IFT UA

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Content-Length: 0
```

F4 100 Trying IFTGW UA -> PROXY

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: sip:+1-650-555-2222@ss1.wcom.com;user=phone
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Content-Length: 0
```

F5 180 Ringing IFTGW UA -> PROXY

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Content-Length: 0
```

F6 180 Ringing PROXY -> IFT UA

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
```

```
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Content-Length: 0
```

F7 200 OK IFTGW UA -> PROXY

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com;user=phone>
Content-Type: application/sdp
Content-Length: 150

v=0
o=IFAXTERMINAL01 2890844527 2890844527 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12323 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F8 200 OK PROXY -> IFT UA

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com;
               maddr=ss1.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 17 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com;user=phone>
Content-Type: application/sdp
Content-Length: 150

v=0
o=IFAXTERMINAL01 2890844527 2890844527 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=audio 12323 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F9 ACK IFT UA -> PROXY

```
ACK sip:+1-650-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ift.here.com:5060
Route: <sip:+1-650-555-2222@iftgw.there.com;maddr=ss1.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 17 ACK
Content-Length: 0
```

F10 ACK PROXY -> IFTGW UA

```
ACK sip:+1-650-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
Record-Route: <sip:+1-650-555-2222@iftgw.there.com;
 maddr=ss1.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 17 ACK
Content-Length: 0
```

/* RTP streams are established. The CNG fax tone is sent in-band if it is present. The receiving side IFT gateway DSP detects the Preamble. A new UDP port is open on IFTGW for T.38 IFP packets and the IFTGW signals the switch over to fax mode by send a re-INVITE with the same Call-ID and with the new UDP port in the SDP */

F11 INVITE IFTGW UA -> PROXY

```
INVITE sip:+1-303-555-1111@ift.here.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
To: sip:+1-303-555-1111@ift.here.com;user=phone
From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 1 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com;user=phone>
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890844527 2890844527 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=image 49172 udptl t38
a=T38FaxVersion:0
```

```
   a=T38maxBitRate:14400
   a=T38FaxFillBitRemoval:0
   a=T38FaxTranscodingMMR:0
   a=T38FaxTranscodingJBIG:0
   a=T38FaxRateManagement:transferredTCF
   a=T38FaxMaxBuffer:72
   a=T38FaxMaxDatagram:316
   a=T38FaxUdpEC:t38UDPRedundancy



   F12 INVITE  PROXY -> IFT UA

   INVITE sip:+1-303-555-1111@ift.here.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d008.1
   Via: SIP/2.0/UDP iftgw.there.com:5060
   Record-Route: <sip:+1-650-555-2222@ss1.wcom.com;
                 maddr=ss1.wcom.com>
   To: sip:+1-303-555-1111@ift.here.com;user=phone
   From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
   Call-ID: 1717@ift.here.com
   CSeq: 1 INVITE
   Contact: <sip:+1-650-555-2222@iftgw.there.com;user=phone>
   Content-Type: application/sdp
   Content-Length: 320

   v=0
   o=faxgw1 2890844527 2890844527 IN IP4 iftgw.there.com
   s=Session SDP
   c=IN IP4 iftmg.there.com
   t=0 0
   m=image 49172 udptl t38
   a=T38FaxVersion:0
   a=T38maxBitRate:14400
   a=T38FaxFillBitRemoval:0
   a=T38FaxTranscodingMMR:0
   a=T38FaxTranscodingJBIG:0
   a=T38FaxRateManagement:transferredTCF
   a=T38FaxMaxBuffer:72
   a=T38FaxMaxDatagram:316
   a=T38FaxUdpEC:t38UDPRedundancy


   F13 200 OK IFT UA -> PROXY

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d008.1
   Via: SIP/2.0/UDP iftgw.there.com:5060
   To: sip:+1-303-555-1111@ift.here.com;user=phone
   From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
   Call-ID: 1717@ift.here.com
```

```
CSeq: 1 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com;user=phone>
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890846527 2890846527 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=image 15002 udptl t38
a=T38FaxVersion:0
a=T38maxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPRedundancy
```

F14 200 OK PROXY -> IFT UA

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP iftgw.there.com:5060
Record-Route: <sip:+1-650-555-2222@ss1.wcom.com;
              maddr=ss1.wcom.com>
To: sip:+1-303-555-1111@ift.here.com;user=phone
From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 1 INVITE
Contact: <sip:+1-303-555-1111@ift.here.com;user=phone>
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890846527 2890846527 IN IP4 ift.here.com
s=Session SDP
c=IN IP4 iftmg.here.com
t=0 0
m=image 15002 udptl t38
a=T38FaxVersion:0
a=T38maxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:72
a=T38FaxMaxDatagram:316
a=T38FaxUdpEC:t38UDPRedundancy
```

F15 ACK IFTGW UA -> PROXY

```
ACK sip:+1-303-555-1111@ift.here.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
To: sip:+1-303-555-1111@ift.here.com;user=phone
From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 1 ACK
Content-Length: 0
```

F16 ACK PROXY -> IFT UA

```
ACK sip:+1-303-555-1111@ift.here.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d008.1
Via: SIP/2.0/UDP iftgw.there.com:5060
To: sip:+1-303-555-1111@ift.here.com;user=phone
From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 1 ACK
Content-Length: 0
```

/* T.38 fax transmission established both ways which replaces the PCM audio session. */

/* Then, the end of the fax transmission is detected on ingress side and sent to the egress side (IFTGW). IFTGW initiates the switch back to voice communication */

F17 INVITE IFTGW UA -> PROXY

```
INVITE sip:+1-303-555-1111@ift.here.com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
Route: <sip:+1-650-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
To: sip:+1-303-555-1111@ift.here.com;user=phone
From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 2 INVITE
Contact: <sip:+1-650-555-2222@iftgw.there.com;user=phone>
Content-Type: application/sdp
Content-Length: 181

v=0
o=faxgw1 2890844527 2890844527 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
```

```
   m=audio 12323 RTP/AVP 0
   a=rtpmap:0 PCMU/8000


   F18 INVITE  PROXY -> IFT UA

   INVITE sip:+1-303-555-1111@ift.here.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d009.1
   Via: SIP/2.0/UDP iftgw.there.com:5060
   Record-Route: <sip:+1-650-555-2222@ss1.wcom.com;
                 maddr=ss1.wcom.com>
   To: sip:+1-303-555-1111@ift.here.com;user=phone
   From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
   Call-ID: 1717@ift.here.com
   CSeq: 2 INVITE
   Contact: <sip:+1-650-555-2222@iftgw.there.com;user=phone>
   Content-Type: application/sdp
   Content-Length: 181

   v=0
   o=faxgw1 2890844527 2890844527 IN IP4 iftgw.there.com
   s=Session SDP
   c=IN IP4 iftmg.there.com
   t=0 0
   m=audio 12323 RTP/AVP 0
   a=rtpmap:0 PCMU/8000


   F19 200 OK IFT UA -> PROXY

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d008.1
   Via: SIP/2.0/UDP iftgw.there.com:5060
   Record-Route: <sip:+1-650-555-2222@ss1.wcom.com;
                 maddr=ss1.wcom.com>
   To: sip:+1-303-555-1111@ift.here.com;user=phone
   From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
   Call-ID: 1717@ift.here.com
   CSeq: 2 INVITE
   Contact: <sip:+1-303-555-1111@ift.here.com;user=phone>
   Content-Type: application/sdp
   Content-Length: 150

   v=0
   o=faxgw1 2890844527 2890844527 IN IP4 ift.here.com
   s=Session SDP
   c=IN IP4 iftmg.here.com
   t=0 0
   m=audio 3456 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

```
   F20 200 OK  PROXY -> IFTGW UA

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP iftgw.there.com:5060
   Record-Route: <sip:+1-650-555-2222@ss1.wcom.com;
                 maddr=ss1.wcom.com>
   To: sip:+1-303-555-1111@ift.here.com;user=phone
   From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
   Call-ID: 1717@ift.here.com
   CSeq: 2 INVITE
   Contact: <sip:+1-303-555-1111@ift.here.com;user=phone>
   Content-Type: application/sdp
   Content-Length: 150

   v=0
   o=faxgw1 2890844527 2890844527 IN IP4 ift.here.com
   s=Session SDP
   c=IN IP4 iftmg.here.com
   t=0 0
   m=audio 3456 RTP/AVP 0
   a=rtpmap:0 PCMU/8000


   F21 ACK IFTGW UA -> PROXY

   ACK sip:+1-303-555-1111@ift.here.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP iftgw.there.com:5060
   Route: <sip:+1-650-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
   To: sip:+1-303-555-1111@ift.here.com;user=phone
   From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
   Call-ID: 1717@ift.here.com
   CSeq: 2 ACK
   Content-Length: 0


   F22 ACK  PROXY -> IFT UA

   ACK sip:+1-303-555-1111@ift.here.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d009.1
   Via: SIP/2.0/UDP iftgw.there.com:5060
   To: sip:+1-303-555-1111@ift.here.com;user=phone
   From: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
   Call-ID: 1717@ift.here.com
   CSeq: 2 ACK
   Content-Length: 0


   F23 BYE IFT UA -> PROXY

   BYE sip:+1-650-555-2222@ss1.wcom.com SIP/2.0
```

```
Via: SIP/2.0/UDP ift.here.com:5060
Route: <sip:+1-650-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 18 BYE
Content-Length: 0


F24 BYE PROXY -> IFTGW UA

BYE sip:+1-650-555-2222@ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d009.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 18 BYE
Content-Length: 0


F25 200 OK IFTGW UA -> PROXY

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060; branch=2d007.1
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 18 BYE
Content-Type: application/sdp
Content-Length: 0


F26 200 OK  PROXY -> IFT UA

SIP/2.0 200 OK
Via: SIP/2.0/UDP ift.here.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: <sip:+1-650-555-2222@ss1.wcom.com;user=phone>;tag=5617
Call-ID: 1717@ift.here.com
CSeq: 18 BYE
Content-Type: application/sdp
Content-Length: 0
```

3.2 Failure Scenarios

3.2.1 Unsuccessful SIP to SIP no answer

```
   User A          Proxy 1          Proxy 2          User B
     |                |                |                |
     |   INVITE F1    |                |                |
     |--------------->|   INVITE F2    |                |
     |    (100) F3    |--------------->|   INVITE F4    |
     |<---------------|    (100) F5    |--------------->|
     |                |<---------------|                |
     |                |                |      180 F6    |
     |                |     180 F7     |<---------------|
     |     180 F8     |<---------------|                |
     |<---------------|                |                |
     |                |                |                |
     |   CANCEL F9    |                |                |
     |--------------->|                |                |
     |     200 F10    |                |                |
     |<---------------|   CANCEL F11   |                |
     |                |--------------->|                |
     |                |     200 F12    |                |
     |                |<---------------|                |
     |                |                |   CANCEL F13   |
     |                |                |--------------->|
     |                |                |     200 F14    |
     |                |                |<---------------|
     |                |                |     487 F15    |
     |                |                |<---------------|
     |                |                |     ACK F16    |
     |                |     487 F17    |--------------->|
     |                |<---------------|                |
     |                |     ACK F18    |                |
     |     487 F19    |--------------->|                |
     |<---------------|                |                |
     |     ACK F20    |                |                |
     |--------------->|                |                |
     |                |                |                |
```

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 CANCEL message is acknowledged with a 200 OK on a hop by hop basis, rather than end to end.

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
 nonce="ze7k1ee88df84f1cec431ae6cbe5a359", opaque="",
 uri="sip:ss1.wcom.com", response="b00b416324679d7e243f55708d44be7b"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/*Client for A prepares to receive data on port 49172 from the network.*/

F2 INVITE Proxy 1 -> Proxy 2

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
```

```
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F3 (100 Trying) Proxy 1 -> A

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

F4 INVITE Proxy 2 -> B

```
INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F5 (100 Trying) Proxy 2 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 180 Ringing B -> Proxy 2

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F7 180 Ringing Proxy 2 -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F8 180 Ringing Proxy 1 -> A

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

F9 CANCEL A -> Proxy 1

```
CANCEL sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0
```

F10 200 OK Proxy 1 -> A

```
SIP/2.0 200 OK
```

```
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 12345600@here.com
   CSeq: 1 CANCEL
   Content-Length: 0


   F11 CANCEL Proxy 1 -> Proxy 2

   CANCEL sip:UserA@here.com SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 12345600@here.com
   CSeq: 1 CANCEL
   Content-Length: 0


   F12 200 OK Proxy 2 -> Proxy 1

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 12345600@here.com
   CSeq: 1 CANCEL
   Content-Length: 0


   F13 CANCEL Proxy 2 -> B

   CANCEL sip:UserB@110.111.112.113 SIP/2.0
   Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 12345600@here.com
   CSeq: 1 CANCEL
   Content-Length: 0


   F14 200 OK B -> Proxy 2

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>
   Call-ID: 12345600@here.com
   CSeq: 1 CANCEL
   Content-Length: 0
```

F15 487 Request Cancelled B -> Proxy 2

```
SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F16 ACK Proxy 2 -> B

```
ACK sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

F17 487 Request Cancelled Proxy 2 -> Proxy 1

```
SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=9876
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F18 ACK Proxy 1 -> Proxy 2

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=9876
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

F19 487 Request Cancelled Proxy 1 -> A

```
SIP/2.0 487 Request Cancelled
```

```
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=4321
Call-ID: 12345600@here.com
CSeq: 1 INVITE
```

F20 ACK A -> Proxy 1

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=4321
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

3.2.2 Unsuccessful SIP to SIP busy

```
   User A          Proxy 1          Proxy 2          User B
     |                |                |                |
     |   INVITE F1    |                |                |
     |--------------->|   INVITE F2    |                |
     |    (100) F3    |--------------->|   INVITE F4    |
     |<---------------|    (100) F5    |--------------->|
     |                |<---------------|                |
     |                |                |     486 F6     |
     |                |                |<---------------|
     |                |                |     ACK F7     |
     |                |     486 F8     |--------------->|
     |                |<---------------|                |
     |                |     ACK F9     |                |
     |     486 F10    |--------------->|                |
     |<---------------|                |                |
     |     ACK F11    |                |                |
     |--------------->|                |                |
     |                |                |                |
```

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

Message Details

F1 INVITE User A -> Proxy 1

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
 nonce="dc3a5ab2530aa93112cf5904ba7d88fa", opaque="",
 uri="sip:ss1.wcom.com", response="702138b27d869ac8741e10ec643d55be"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
```

```
a=rtpmap:0 PCMU/8000


/*Client for A prepares to receive data on port 49172 from the
network.*/

F2 INVITE Proxy 1 -> Proxy 2

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F3 (100 Trying) Proxy 1 -> User A

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


F4 INVITE Proxy 2 -> User B

INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
```

```
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F5 (100 Trying) Proxy 2 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 486 Busy Here User B -> Proxy 2

```
SIP/2.0  486 Busy Here
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F7 ACK Proxy 2 -> User B

```
ACK sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

F8 486 Busy Here Proxy 2 -> Proxy 1

```
SIP/2.0  486 Busy Here
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=1293
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F9 ACK Proxy 1 -> Proxy 2

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=1293
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

F10 486 Busy Here Proxy 1 -> User A

```
SIP/2.0  486 Busy Here
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=a6b4
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F11 ACK User A -> Proxy 1

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=a6b4
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

3.2.3 Unsuccessful SIP to SIP no response

```
   User A          Proxy 1          Proxy 2          User B
      |                |                 |                 |
      |   INVITE F1    |                 |                 |
      |--------------->|   INVITE F2     |                 |
      |    (100) F3    |---------------->|   INVITE F4     |
      |<---------------|    (100) F5     |---------------->|
      |                |<----------------|   INVITE F6     |
      |                |                 |---------------->|
      |                |                 |   INVITE F7     |
      |                |                 |---------------->|
      |                |                 |   INVITE F8     |
      |                |                 |---------------->|
      |                |                 |   INVITE F9     |
      |                |                 |---------------->|
      |                |                 |   INVITE F10    |
      |                |                 |---------------->|
      |                |                 |   INVITE F11    |
      |                |                 |---------------->|
      |                |                 |   CANCEL F12    |
      |                |     480 F13     |---------------->|
      |                |<----------------|                 |
      |                |     ACK F14     |                 |
      |     480 F15    |---------------->|                 |
      |<---------------|                 |                 |
      |     ACK F16    |                 |                 |
      |--------------->|                 |                 |
      |                |                 |                 |
```

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 Call Flow 5.2.6.

Message Details

F1 INVITE User A -> Proxy 1

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
```

```
 realm="MCI WorldCom SIP",
 nonce="cf5904ba7d8dc3a5ab2530aa931128fa", opaque="",
 uri="sip:ss1.wcom.com", response="7afc04be7961f053c24f80e7dbaf888f"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/*Client for A prepares to receive data on port 49172 from the network.*/

F2 INVITE Proxy 1 -> Proxy 2

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F3 (100 Trying) Proxy 1 -> User A

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

F4 INVITE Proxy 2 -> User B

```
INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F5 (100 Trying) Proxy 2 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 INVITE Proxy 2 -> User B

Resend of Message F4

F7 INVITE Proxy 2 -> User B

Resend of Message F4

F8 INVITE Proxy 2 -> User B

Resend of Message F4

F9 INVITE Proxy 2 -> User B

Resend of Message F4

F10 INVITE Proxy 2 -> User B

Resend of Message F4

F11 INVITE Proxy 2 -> User B

Resend of Message F4

F12 CANCEL Proxy 2 -> User B

```
CANCEL sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0
```

F13 480 No Response Proxy 2 -> Proxy 1

```
SIP/2.0  480 No Response
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F14 ACK Proxy 1 -> Proxy 2

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

F15 480 No Response Proxy 1 -> User A

```
SIP/2.0  480 No Response
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F16 ACK User A -> Proxy 1

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

3.2.4 Unsuccessful SIP to SIP Temporarily Unavailable

```
   User A          Proxy 1          Proxy 2          User B
     |                |                |                |
     |   INVITE F1    |                |                |
     |--------------->|   INVITE F2    |                |
     |    (100) F4    |--------------->|   INVITE F3    |
     |<---------------|    (100) F5    |--------------->|
     |                |<---------------|    (100) F6    |
     |                |                |<---------------|
     |                |                |      180 F7    |
     |                |     180 F8     |<---------------|
     |     180 F9     |<---------------|                |
     |<---------------|                |     480 F10    |
     |                |                |<---------------|
     |                |                |     ACK F11    |
     |                |     480 F12    |--------------->|
     |                |<---------------|                |
     |                |     ACK F13    |                |
     |     480 F14    |--------------->|                |
     |<---------------|                |                |
     |     ACK F15    |                |                |
     |--------------->|                |                |
     |                |                |                |
```

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

F1 INVITE A -> Proxy 1

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
 nonce="aa9311cf5904ba7d8dc3a5ab253028fa", opaque="",
 uri="sip:ss1.wcom.com", response="59a46a91bf1646562a4d486c84b399db"
Content-Type: application/sdp
Content-Length: 147
```

```
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/*Client for A prepares to receive data on port 49172 from the network.*/

F2 INVITE Proxy 1 -> Proxy 2

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F3 INVITE Proxy 2 -> B

```
INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss2.wcom.com>,
 <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147
```

```
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) Proxy 1 -> A

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

F5 (100 Trying) Proxy 2 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 (100 Trying) User B -> Proxy 2

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F7 180 Ringing B -> Proxy 2

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
```

```
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F8 180 Ringing Proxy 2 -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F9 180 Ringing Proxy 1 -> A

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

F10 480 Temporarily Unavailable B -> Proxy 2

```
SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F11 ACK Proxy 2 -> B

```
ACK sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss2.wcom.com:5060;branch=721e418c4.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
```

```
   Content-Length: 0


   F12 480 Temporarily Unavailable Proxy 2 -> Proxy 1

   SIP/2.0 480 Temporarily Unavailable
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=9
   Call-ID: 12345600@here.com
   CSeq: 1 INVITE
   Content-Length: 0


   F13 ACK Proxy 1 -> Proxy 2

   ACK sip:UserB@there.com SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=9
   Call-ID: 12345600@here.com
   CSeq: 1 ACK
   Content-Length: 0


   F14 480 Temporarily Unavailable Proxy 1 -> A

   SIP/2.0 480 Temporarily Unavailable
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=489292845645245422
   Call-ID: 12345600@here.com
   CSeq: 1 INVITE
   Content-Length: 0


   F15 ACK A -> Proxy 1

   ACK sip:UserB@there.com SIP/2.0
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:UserA@here.com>
   To: LittleGuy <sip:UserB@there.com>;tag=489292845645245422
   Call-ID: 12345600@here.com
   CSeq: 1 ACK
   Content-Length: 0
```

4 SIP to Gateway Dialing

In the following scenarios, User A (BigGuy 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 Proxy 1 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, since the call could be redirected back to the SIP network.

There is a major difference in the call flows in this section. 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 responses used throughout this document in place of the 180 Ringing.

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 Proxy 1 and a Gateway.

4.1.1 Successful SIP to ISUP PSTN call

```
   User A          Proxy 1           NGW 1           User B
     |                |                 |                |
     |   INVITE F1    |                 |                |
     |--------------->|                 |                |
     |    (100) F2    |                 |                |
     |<---------------|   INVITE F3     |                |
     |                |---------------->|                |
     |                |    (100) F4     |                |
     |                |<----------------|    IAM F5      |
     |                |                 |--------------->|
     |                |                 |    ACM F6      |
     |                |     183 F7      |<---------------|
     |     183 F8     |<----------------|                |
     |<---------------|                 |                |
     |        One Way RTP Media         |  One Way Voice |
     |<=================================|<===============|
     |                |                 |     ANM F9     |
     |                |    200 F10      |<---------------|
     |     200 F11    |<----------------|                |
     |<---------------|                 |                |
     |     ACK F12    |                 |                |
     |--------------->|     ACK F13     |                |
     |                |---------------->|                |
     |       Both Way RTP Media         | Both Way Voice |
     |<================================>|<==============>|
     |     BYE F14    |                 |                |
     |--------------->|     BYE F15     |                |
     |                |---------------->|                |
     |                |     200 F16     |                |
     |     200 F17    |<----------------|    REL F18     |
     |<---------------|                 |--------------->|
     |                |                 |    RLC F19     |
     |                |                 |<---------------|
     |                |                 |                |
```

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 (NGW 1) 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 NGW 1 and User B's telephone switch is ANSI ISUP.

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:+1-972-555-2222@ss1.wcom.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
 nonce="dc3a5ab25302aa931904ba7d88fa1cf5", opaque="",
 uri="sip:ss1.wcom.com", response="ccdca50cb091d587421457305d097458c"
Content-Type: application/sdp
Content-Length: 140

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 (100 Trying) Proxy 1 -> User A

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

/* Proxy 1 uses a Location Service function to determine where B is located. Based upon location analysis the call is forwarded to NGW 1. Client for A prepares to receive data on port 49172 from the network.*/

F3 INVITE Proxy 1 -> NGW 1

```
INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 140

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) NGW 1 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 IAM NGW 1 -> 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
```

F6 ACM User B -> NGW 1

```
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


F7 183 Session Progress NGW 1 -> Proxy 1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0


/* NGW 1 sends PSTN audio (ringing) in the RTP path to A */

F8 183 Session Progress Proxy 1 -> User A

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0


F9 ANM User B -> NGW 1

ANM


F10 200 OK NGW 1 -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <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: 164

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

F11 200 OK Proxy 1 -> User A

```
   SIP/2.0 200 OK
   Via: SIP/2.0/UDP here.com:5060
   Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
   From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
   To: LittleGuy <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: 164

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

F12 ACK A -> Proxy 1

```
   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: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
   To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
    ;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 1 ACK
   Content-Length: 0
```

F13 ACK Proxy 1 -> NGW 1

```
ACK sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <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 NGW 1) */

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

F14 BYE A -> Proxy 1

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: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0


F15 BYE Proxy 1 -> NGW 1

BYE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0


F16 200 OK NGW 1 -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
```

```
Content-Length: 0


F17 200 OK Proxy 1 -> A

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


F18 REL NGW 1 -> B

REL
CauseCode=16 Normal
CodingStandard=CCITT


F19 RLC B -> NGW 1

RLC
```

4.1.2 Successful SIP to ISDN PBX call

```
   User A          Proxy 1          GW 1            PBX C
     |                |                |                |
     |   INVITE F1    |                |                |
     |--------------->|                |                |
     |    (100) F2    |                |                |
     |<---------------|   INVITE F3    |                |
     |                |--------------->|                |
     |                |    (100) F4    |                |
     |                |<---------------|    SETUP F5    |
     |                |                |--------------->|
     |                |                |  CALL PROC F6  |
     |                |                |<---------------|
     |                |                |  PROGress F7  |
     |                |    183 F8      |<---------------|
     |    183 F9      |<---------------|                |
     |<---------------|                |                |
     |         One Way RTP Media       |  One Way Voice |
     |<================================|<===============|
     |                |                |   CONNect F10  |
     |                |                |<---------------|
     |                |                | CONNect ACK F11|
     |                |    200 F12     |--------------->|
     |     200 F13    |<---------------|                |
     |<---------------|                |                |
     |     ACK F14    |                |                |
     |--------------->|     ACK F15    |                |
     |                |--------------->|                |
     |        Both Way RTP Media       | Both Way Voice |
     |<===============================>|<==============>|
     |     BYE F16    |                |                |
     |--------------->|     BYE F17    |                |
     |                |--------------->|                |
     |                |     200 F18    |                |
     |     200 F19    |<---------------| DISConnect F20 |
     |<---------------|                |--------------->|
     |                |                |   RELease F21  |
     |                |                |<---------------|
     |                |                | RELease COM F22|
     |                |                |--------------->|
     |                |                |                |
```

User A is a SIP device while User C is connected via an Enterprise Gateway (GW 1) to a PBX. The PBX connection is via a ISDN trunk group. User A dials User C’s telephone number (918-555-3333) which is globalized and put into a SIP URL.

The host portion of the Request-URI in the INVITE F6 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 by GW 1 and the context of the digits could become lost and the call unroutable.

Proxy 1 looks up the telephone number and locates the Enterprise Gateway that serves User C. User C is identified by its extension (444-3333) in the Request-URI sent to GW 1.

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

F1 INVITE A -> Proxy 1

```
INVITE sip:+1-918-555-3333@ss1.wcom.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
 nonce="qo0dc3a5ab22aa931904badfa1cf5j9h", opaque="",
 uri="sip:ss1.wcom.com", response="6c792f5c9fa360358b93c7fb826bf550"
Content-Type: application/sdp
Content-Length: 140

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 (100 Trying) Proxy 1 -> User A

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0
```

F3 INVITE Proxy 1 -> GW 1

```
INVITE sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-918-555-3333@ss1.wcom.com;maddr=ss1.wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 140

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) GW -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0
```

F5 SETUP GW 1 -> User C

```
Protocol discriminator=Q.931
Call reference: Flag=0
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 unknown
Digits=444-3333
```

F6 CALL PROCeeding User C -> GW 1

```
Protocol discriminator=Q.931
Call reference: Flag=1
Message type=CALL PROC
Channel identification=Exclusive B-channel
```

F7 PROGress User C -> GW 1

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

F8 183 Session Progress GW 1 -> Proxy 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0
```

/* GW 1 will encode PSTN audio (ringing) to A in RTP path */

F9 183 Session Progress Proxy 1 -> User A

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0
```

F10 CONNect User C -> GW 1

```
Protocol discriminator=Q.931
Call reference: Flag=1
Message type=CONN
```

F11 CONNect ACK GW 1 -> User C

```
   Protocol discriminator=Q.931
   Call reference: Flag=0
   Message type=CONN ACK


   F12 200 OK GW 1 -> Proxy 1

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP here.com:5060
   Record-Route: <sip:+1-918-555-3333@ss1.wcom.com;maddr=ss1.wcom.com>
   From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
   To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
    ;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 2 INVITE
   Contact: <sip:444-3333@wcom.com>
   Content-Type: application/sdp
   Content-Length: 165

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


   F13 200 OK Proxy 1 -> User A

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP here.com:5060
   Record-Route: <sip:+1-918-555-3333@ss1.wcom.com;maddr=ss1.wcom.com>
   From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
   To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 2 INVITE
   Contact: <sip:444-3333@wcom.com>
   Content-Type: application/sdp
   Content-Length: 165

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

F14 ACK A -> Proxy 1

```
ACK sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:444-3333@wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 ACK
Content-Length: 0
```

F15 ACK Proxy 1 -> GW 1

```
ACK sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 ACK
Content-Length: 0
```

/* RTP streams are established between A and B (via GW 1) */

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

F16 BYE A -> Proxy 1

```
BYE sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:444-3333@wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 3 BYE
Content-Length: 0
```

F17 BYE Proxy 1 -> GW 1

```
BYE sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
 ;tag=314159
```

```
Call-ID: 12345600@here.com
CSeq: 3 BYE
Content-Length: 0
```

F18 200 OK GW 1 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 3 BYE
Content-Length: 0
```

F19 200 OK Proxy 1 -> A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: OtherGuy <sip:+1-918-555-3333@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 3 BYE
Content-Length: 0
```

F20 DISConnect GW 1 -> User C

```
Protocol discriminator=Q.931
Call reference: Flag=1
Message type=DISC
Cause=16 (Normal clearing)
```

F21 RELease User C -> GW 1

```
Protocol discriminator=Q.931
Call reference: Flag=0
Message type=REL
```

F22 RELease COMplete GW 1 -> User C

```
Protocol discriminator=Q.931
Call reference: Flag=1
Message type=REL COM
```

4.1.3 Successful SIP to ISUP PSTN call with overflow

```
     User A        Proxy 1          NGW 1          NGW 2          User B
    |              |              |              |              |
    |  INVITE F1   |              |              |              |
    |------------->|              |              |              |
    |              |  INVITE F2   |              |              |
    |   (100) F3   |------------->|              |              |
    |<-------------|    503 F4    |              |              |
    |              |<-------------|              |              |
    |              |    ACK F5    |              |              |
    |              |------------->|              |              |
    |              |   INVITE F6                 |              |
    |              |---------------------------->|     IAM F7   |
    |              |                             |------------->|
    |              |                             |     ACM F8   |
    |              |            183 F9           |<-------------|
    |   183 F10    |<----------------------------|              |
    |<-------------|                             |              |
    |               One Way RTP Media            | One Way Voice|
    |<===========================================|<=============|
    |              |                             |    ANM F11   |
    |              |           200 F12           |<-------------|
    |    200 F13   |<----------------------------|              |
    |<-------------|                             |              |
    |    ACK F14   |                             |              |
    |------------->|            ACK F15          |              |
    |              |---------------------------->|              |
    |             Both Way RTP Media             |Both Way Voice|
    |<==========================================>|<============>|
    |    BYE F16   |                             |              |
    |------------->|           BYE F17           |              |
    |              |---------------------------->|              |
    |              |           200 F18           |              |
    |    200 F19   |<----------------------------|    REL F20   |
    |<-------------|                             |------------->|
    |              |                             |    RLC F21   |
    |              |                             |<-------------|
    |              |                             |              |
```

User A calls User B through Proxy 1. Proxy 1 tries to route to a Network Gateway NGW 1. NGW 1 is not available and responds with a 503 Service Unavailable (F4). The call is then routed to Network Gateway NGW 2. User B answers the call. The call is terminated when User A disconnects the call. NGW 2 and User B’s telephone switch use ANSI ISUP signaling.

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:+1-972-555-2222@ss1.wcom.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
 nonce="b59311c3ba05b401cf80b2a2c5ac51b0", opaque="",
 uri="sip:ss1.wcom.com", response="ba6ab44923fa2614b28e3e3957789ab0"
Content-Type: application/sdp
Content-Length: 140

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine where B is located. Proxy 1 receives a primary route NGW 1 and a secondary route NGW 2. NGW 1 is tried first */

F2 INVITE Proxy 1 -> NGW 1

```
INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 140

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F3 (100 Trying) Proxy 1 -> User A

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F4 503 Service Unavailable NGW 1 -> Proxy 1

```
SIP/2.0 503 Service Unavailable
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=123456789
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 ACK Proxy 1 -> NGW 1

```
ACK sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com>;user=phone>
 ;tag=123456789
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

/* Proxy 1 now tries secondary route to NGW 2 */

F6 INVITE Proxy 1 -> NGW 2

```
INVITE sip:+1-972-555-2222@ngw2.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
```

```
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F7 IAM NGW 2 -> 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
```

F8 ACM User B -> NGW 2

```
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
```

F9 183 Session Progress NGW 2 -> Proxy 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

/* RTP packets are sent by GW to A for audio (i.e. ring tone) */

F10 183 Session Progress Proxy 1 -> User A

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F11 ANM User B -> NGW 2

```
ANM
```

F12 200 OK NGW 2 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <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: 164

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

F13 200 OK Proxy 1 -> User A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <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: 164

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


F14 ACK A -> Proxy 1

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: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0


F15 ACK Proxy 1 -> NGW 2

ACK sip:+1-972-555-2222@ngw2.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <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. */

F16 BYE A -> Proxy 1

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@ngw2.wcom.com;user=phone>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
```

```
    ;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 2 BYE
   Content-Length: 0
```

F17 BYE Proxy 1 -> NGW 2

```
   BYE sip:+1-972-555-2222@ngw2.wcom.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
   To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
    ;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 2 BYE
   Content-Length: 0
```

F18 200 OK NGW 2 -> Proxy 1

```
   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
   To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
    ;tag=314159
   Call-ID: 12345600@here.com
   CSeq: 2 BYE
   Content-Length: 0
```

F19 200 OK Proxy 1 -> User A

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

F20 REL NGW 2 -> B

```
   REL
   CauseCode=16 Normal
   CodingStandard=CCITT
```

F21 RLC B -> NGW 2

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 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          Proxy 1           NGW 1           User B
     |                |                 |                |
     |   INVITE F1    |                 |                |
     |--------------->|                 |                |
     |    (100) F2    |                 |                |
     |<---------------|   INVITE F3     |                |
     |                |---------------->|                |
     |                |    (100) F4     |                |
     |                |<----------------|     IAM F5     |
     |                |                 |--------------->|
     |                |                 |     ACM F6     |
     |                |     183 F7      |<---------------|
     |     183 F8     |<----------------|                |
     |<---------------|                 |                |
     |         One Way RTP Media        |  One Way Voice |
     |<=================================|<===============|
     |                  Treatment Applied                |
     |<==================================================|
     |   CANCEL F9    |                 |                |
     |--------------->|                 |                |
     |     200 F10    |                 |                |
     |<---------------|   CANCEL F11    |                |
     |                |---------------->|                |
     |                |     200 F12     |                |
     |                |<----------------|     REL F13    |
     |                |                 |--------------->|
     |                |                 |     RLC F14    |
     |                |     487 F15     |<---------------|
     |                |<----------------|                |
     |                |     ACK F16     |                |
     |     487 F17    |---------------->|                |
     |--------------->|                 |                |
     |     ACK F18    |                 |                |
     |--------------->|                 |                |
     |                |                 |                |
```

User A calls User B in the PSTN through a proxy server Proxy 1 and a Network Gateway NGW 1. 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

F1 INVITE A -> Proxy 1

```
INVITE sip:+1-972-555-2222@ss1.wcom.com;user=phone  SIP/2.0
```

```
   Via: SIP/2.0/UDP here.com:5060
   From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
   To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
   Call-ID: 12345600@here.com
   CSeq: 1 INVITE
   Contact: <sip:UserA@100.101.102.103>
   Proxy-Authorization:Digest username="UserA",
    realm="MCI WorldCom SIP",
    nonce="01cf8311c3b0b2a2c5ac51bb59a05b40", opaque="",
    uri="sip:ss1.wcom.com", response="e178fbe430e6680a1690261af8831f40"
   Content-Type: application/sdp
   Content-Length: 140

   v=0
   o=UserA 2890844526 2890844526 IN IP4 here.com
   s=Session SDP
   c=IN IP4 here.com
   t=0 0
   m=audio 49172 RTP/AVP 0
   a=rtpmap:0 PCMU/8000


   F2 (100 Trying) Proxy 1 -> A

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


   /* Proxy 1 uses a Location Service function to determine where B is
   located.  Based upon location analysis the call is forwarded to NGW
   1.  Client for A prepares to receive data on port 49172 from the
   network. */

   F3 INVITE Proxy 1 -> NGW 1

   INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP here.com:5060
   Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
   From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
   To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
   Call-ID: 12345600@here.com
   CSeq: 1 INVITE
   Contact: <sip:UserA@100.101.102.103>
   Content-Type: application/sdp
   Content-Length: 140
```

```
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) NGW 1 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 IAM NGW 1 -> 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
```

F6 ACM User B -> NGW 1

```
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
```

F7 183 Session Progress NGW 1 -> Proxy 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
```

```
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F8 183 Session Progress Proxy 1 -> User A

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F9 CANCEL A -> Proxy 1

```
CANCEL sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0
```

F10 200 OK Proxy 1 -> A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0
```

F11 CANCEL Proxy 1 -> NGW 1

```
CANCEL sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0
```

F12 200 OK NGW 1 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0
```

F13 REL NGW 1 -> B

```
REL
CauseCode=16 Normal
CodingStandard=CCITT
```

F14 RLC B -> NGW 1

```
RLC
```

F15 487 Request Cancelled NGW 1 -> Proxy 1

```
SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F16 ACK Proxy 1 -> NGW 1

```
ACK sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

F17 487 Request Cancelled Proxy 1 -> A

```
SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=4f-3e-ff-23-09-43
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F18 ACK A -> Proxy 1

```
ACK sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=4f-3e-ff-23-09-43
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

4.2.2 Unsuccessful SIP to PSTN: REL w/Cause from PSTN

```
   User A           Proxy 1           NGW 1            User B
     |                 |                 |                 |
     |   INVITE F1     |                 |                 |
     |---------------->|                 |                 |
     |    (100) F2     |                 |                 |
     |<----------------|   INVITE F3     |                 |
     |                 |---------------->|                 |
     |                 |    (100) F4     |                 |
     |                 |<----------------|     IAM F5      |
     |                 |                 |---------------->|
     |                 |                 |   REL(28) F6    |
     |                 |                 |<----------------|
     |                 |                 |     RLC F7      |
     |                 |     484 F8      |---------------->|
     |                 |<----------------|                 |
     |                 |     ACK F9      |                 |
     |                 |---------------->|                 |
     |     484 F10     |                 |                 |
     |<----------------|                 |                 |
     |     ACK F11     |                 |                 |
     |---------------->|                 |                 |
     |                 |                 |                 |
```

User A calls PSTN User B through a Proxy Server Proxy 1 and a Network Gateway NGW 1. However, User A does not provide enough digits for the call to be completed. The call is rejected by the PSTN with a ANSI ISUP 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. For more details of ISUP cause value to SIP responses refer to [5].

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:+44-1234@ss1.wcom.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+44-1234@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
 nonce="j1c3b0b01cf832da2c5ac51bb59a05b40", opaque="",
 uri="sip:ss1.wcom.com", response="a451358d46b55512863efe1dccaa2f42"
Content-Type: application/sdp
```

```
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2 (100 Trying) Proxy 1 -> A

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+44-1234@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

/* Proxy 1 uses a Location Service function to determine where B is located. Based upon location analysis the call is forwarded to NGW1. Client for A prepares to receive data on port 49172 from the network. */

F3 INVITE Proxy 1 -> NGW 1

```
INVITE sip:+44-1234@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:+44-1234@ss1.wcom.com;maddr=ss1.wcom.com>
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+44-1234@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 here.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) NGW 1 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+44-1234@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 IAM NGW 1 -> User B

```
IAM
CdPN=44-1234,NPI=E.164,NOA=International
CgPN=314-555-1111,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI=Not Required
```

F6 REL User B -> NGW 1

```
REL
CauseValue=28 Address Incomplete
CodingStandard=CCITT
```

F7 RLC NGW 1 -> User B

```
RLC
```

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

F8 484 Address Incomplete NGW 1 -> Proxy 1

```
SIP/2.0 484 Address Incomplete
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+44-1234@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F9 ACK Proxy 1 -> NGW 1

```
ACK sip:+44-1234@ngw1.wcom.com;user=phone SIP/2.0
```

```
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+44-1234@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

F10 484 Address Incomplete Proxy 1 -> User A

```
SIP/2.0 484 Address Incomplete
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+44-1234@ss1.wcom.com;user=phone>;tag=141593
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F11 ACK User A -> Proxy 1

```
ACK sip:+44-1234@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+44-1234@ss1.wcom.com;user=phone>;tag=141593
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

4.2.3 Unsuccessful SIP to PSTN: ANM Timeout

```
   User A          Proxy 1           NGW 1           User B
     |                |                 |                |
     |   INVITE F1    |                 |                |
     |--------------->|                 |                |
     |    (100) F2    |                 |                |
     |<---------------|   INVITE F3     |                |
     |                |---------------->|                |
     |                |    (100) F4     |                |
     |                |<----------------|     IAM F5     |
     |                |                 |---------------->|
     |                |                 |     ACM F6     |
     |                |      183 F7     |<----------------|
     |     183 F8     |<----------------|                |
     |<---------------|                 |                |
     |                |      Timer on NGW 1 Expires      |
     |                |                 |                |
     |                |                 |     REL F9     |
     |                |                 |---------------->|
     |                |                 |    RLC F10     |
     |                |     480 F11     |<----------------|
     |                |<----------------|                |
     |                |     ACK F12     |                |
     |                |---------------->|                |
     |     480 F13    |                 |                |
     |<---------------|                 |                |
     |     ACK F14    |                 |                |
     |--------------->|                 |                |
```

User A calls User B in the PSTN through a proxy server Proxy 1 and Network Gateway NGW 1. The call is released by the Gateway after a timer expires due to no ANswer Message (ANM) being received. The Gateway sends an ISUP Release REL message to the PSTN and a 480 Temporarily Unavailable response to User A in the SIP network.

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:+1-972-555-2222@ss1.wcom.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:UserA@100.101.102.103>
Proxy-Authorization:Digest username="UserA",
 realm="MCI WorldCom SIP",
```

```
     nonce="da2c5ac51bb59a05j1c3b0b01cf832b40", opaque="",
     uri="sip:ss1.wcom.com", response="579cb9db184cdc25bf816f37cbc03c7d"
   Content-Type: application/sdp
   Content-Length: 147

   v=0
   o=UserA 2890844526 2890844526 IN IP4 here.com
   s=Session SDP
   c=IN IP4 here.com
   t=0 0
   m=audio 49172 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine where B is located. Based upon location analysis the call is forwarded to NGW 1. Client for A prepares to receive data on port 49172 from the network.*/

F2 (100 Trying Proxy 1 -> A

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

F3 INVITE Proxy 1 -> NGW 1

```
   INVITE sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP here.com:5060
   Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
   From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
   To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
   Call-ID: 12345600@here.com
   CSeq: 1 INVITE
   Contact: <sip:UserA@100.101.102.103>
   Content-Type: application/sdp
   Content-Length: 147

   v=0
   o=UserA 2890844526 2890844526 IN IP4 here.com
   s=Session SDP
   c=IN IP4 here.com
   t=0 0
   m=audio 49172 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) NGW 1 -> Proxy 1

```
SIP/2.0  100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 IAM NGW 1 -> 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
```

F6 ACM User B -> NGW 1

```
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
```

F7 183 Session Progress NGW 1 -> Proxy 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F8 183 Session Progress Proxy 1 -> User A

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

/* After NGW 1’s timer expires, Network Gateway sends REL to ISUP network and 480 to SIP network */

F9 REL NGW 1 -> User B

```
REL
CauseCode=16 Normal
CodingStandard=CCITT
```

F10 RLC User B -> NGW 1

```
RLC
```

F11 480 Temporarily Unavailable NGW 1 -> Proxy 1

```
SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F12 ACK Proxy 1 -> NGW 1

```
ACK sip:+1-972-555-2222@ngw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
```

F13 480 Temporarily Unavailable F13 Proxy 1 -> User A

```
SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=415913
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
```

F14 ACK User A -> Proxy 1

```
ACK sip:+1-972-555-2222@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:+1-314-555-1111@ss1.wcom.com;user=phone>
To: LittleGuy <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
 ;tag=415913
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 (NGW 1) using ANSI ISUP.

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, NGW 1 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 NGW 1 and played in this speech path. When the call completes successfully, NGW 1 bridges the PSTN speech path with the IP media path. Alternatively, the Gateway could redirect the call to an Announcement Server which would complete the call and play announcements or tones as directed by the Gateway.

5.1.1 Successful PSTN to SIP call

```
   User A           NGW 1          Proxy 1          User B
     |                |                |                |
     |     IAM F1     |                |                |
     |--------------->|   INVITE F2    |                |
     |                |--------------->|   INVITE F3    |
     |                |    (100) F4    |--------------->|
     |                |<---------------|                |
     |                |                |     180 F5     |
     |                |    180 F6      |<---------------|
     |     ACM F7     |<---------------|                |
     |<---------------|                |                |
     |  One Way Voice |                |                |
     |<===============|                |                |
     |  Ringing Tone  |                |     200 F8     |
     |<===============|    200 F9      |<---------------|
     |                |<---------------|                |
     |                |     ACK F10    |                |
     |     ANM F12    |--------------->|    ACK F11     |
     |<---------------|                |--------------->|
     | Both Way Voice |       Both Way RTP Media        |
     |<==============>|<===============================>|
     |     REL F13    |                |                |
     |--------------->|                |                |
     |     RLC F14    |                |                |
     |<---------------|     BYE F15    |                |
     |                |--------------->|    BYE F16     |
     |                |                |--------------->|
     |                |                |    200 F17     |
     |                |     200 F18    |<---------------|
     |                |<---------------|                |
     |                |                |                |
```

In this scenario, User A from the PSTN calls User B through a Network Gateway NGW1 and Proxy Server Proxy 1. 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 an ISUP RELease message which is mapped to a BYE by NGW 1.

Message Details

F1 IAM User A -> NGW 1

```
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


F2 INVITE A -> Proxy 1

INVITE sip:+1-972-555-2222@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:+1-972-555-2222@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


/* Proxy 1 uses a Location Service function to determine where B is
located.  Based upon location analysis the call is forwarded to NGW
1.  NGW 1  prepares to receive data on port 3456 from User A.*/

F3 INVITE Proxy 1 -> User B

INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) User B -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 180 Ringing User B -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 180 Ringing Proxy 1 -> NGW 1

```
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F7 ACM NGW 1 -> 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
```

F8 200 OK User B -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
Contact: <sip:UserB@110.111.112.113>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 150

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

F9 200 OK Proxy 1 -> NGW 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.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 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 150

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

F10 ACK NGW 1 -> Proxy 1

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:UserB@110.111.112.113>
From: <sip:+1-314-555-1111@ngw1.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
```

F11 ACK Proxy 1 -> User B

```
ACK sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0
```

F12 ANM User B -> NGW 1

```
ANM
```

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

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

F13 REL User A -> NGW 1

```
REL
CauseCode=16 Normal
CodingStandard=CCITT
```

F14 RLC NGW 1 -> User A

```
RLC
```

F15 BYE NGW 1-> Proxy 1

```
BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:UserB@110.111.112.113>
From: <sip:+1-314-555-1111@ngw1.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
```

F16 BYE Proxy 1 -> User B

```
BYE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0
```

F17 200 OK User B -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0
```

F18 200 OK Proxy 1 -> NGW 1

```
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@ss1.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

```
   User A           NGW 1           Proxy 1          User B
     |                |                 |                |
     |     IAM F1     |                 |                |
     |--------------->|   INVITE F2     |                |
     |                |---------------->|   INVITE F3    |
     |                |    (100) F4     |--------------->|
     |                |<----------------|                |
     |                |                 |     200 F5     |
     |                |     200 F6      |<---------------|
     |                |<----------------|                |
     |                |     ACK F7      |                |
     |     ANM F9     |---------------->|     ACK F8     |
     |<---------------|                 |--------------->|
     | Both Way Voice |        Both Way RTP Media        |
     |<==============>|<================================>|
     |     REL F10    |                 |                |
     |--------------->|                 |                |
     |     RLC F11    |                 |                |
     |<---------------|     BYE F12     |                |
     |                |---------------->|     BYE F13    |
     |                |                 |--------------->|
     |                |                 |     200 F14    |
     |                |     200 F15     |<---------------|
     |                |<----------------|                |
     |                |                 |                |
```

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). Note that for ETSI and some other ISUP variants, a CONnect message (CON) would be sent instead of the ANM.

Message Details

F1 IAM User A -> NGW 1

```
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
```

F2 INVITE NGW 1 -> Proxy 1

```
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@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine where B is located. Based upon location analysis the call is forwarded to User B. User B prepares to receive data on port 3456 from User A.*/

F3 INVITE Proxy 1 -> User B

```
INVITE UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) Proxy 1 -> NGW 1

```
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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 200 OK User B -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.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 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 150

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

F6 200 OK Proxy 1 -> NGW 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.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 INVITE
Contact: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 150

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

F7 ACK NGW 1 -> Proxy 1

```
ACK UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:UserB@110.111.112.113>
From: <sip:+1-314-555-1111@ngw1.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
```

F8 ACK Proxy 1 -> User B

```
ACK UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0
```

F9 ANM User B -> NGW 1

```
ANM
```

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

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

F10 REL ser A -> NGW 1

```
REL
CauseCode=16 Normal
CodingStandard=CCITT
```

F11 RLC NGW 1 -> User A

```
RLC
```

F12 BYE NGW 1 -> Proxy 1

```
BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:UserB@110.111.112.113>
From: <sip:+1-314-555-1111@ngw1.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
```

F13 BYE Proxy 1 -> User B

```
BYE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0
```

F14 200 OK User B -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 2 BYE
Content-Length: 0
```

F15 200 OK Proxy 1 -> NGW 1

```
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@ss1.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

```
   PBX A            GW 1            Proxy 1           User B
     |                |                 |                 |
     |    Seizure     |                 |                 |
     |--------------->|                 |                 |
     |      Wink      |                 |                 |
     |--------------->|                 |                 |
     |  MF Digits F1  |                 |                 |
     |--------------->|   INVITE F2     |                 |
     |                |---------------->|   INVITE F3     |
     |                |    (100) F4     |---------------->|
     |                |<----------------|                 |
     |                |                 |      180 F5     |
     |                |    180 F6       |<----------------|
     |                |<----------------|                 |
     |  One Way Voice |                 |                 |
     |<===============|                 |                 |
     |  Ringing Tone  |                 |      200 F7     |
     |<===============|     200 F8      |<----------------|
     |                |<----------------|                 |
     |                |     ACK F9      |                 |
     |     Seizure    |---------------->|     ACK F10     |
     |<---------------|                 |---------------->|
     | Both Way Voice |       Both Way RTP Media          |
     |<==============>|<=================================>|
     | Seizure Removal|                 |                 |
     |--------------->|                 |                 |
     | Seizure Removal|                 |                 |
     |<---------------|     BYE F11     |                 |
     |                |---------------->|     BYE F12     |
     |                |                 |---------------->|
     |                |                 |     200 F13     |
     |                |     200 F14     |<----------------|
     |                |<----------------|                 |
     |                |                 |                 |
```

In this scenario, User A dials from PBX A to User B through GW 1 and Proxy 1. This is an example of a call that appears destined for the PSTN but instead is routed to a SIP Client.

Signaling between PBX A and GW 1 is Feature Group B (FGB) circuit associated signaling, in-band Mult-Frequency (MF) outpulsing. After the receipt of the 180 Ringing from User B, GW 1 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.

The Enterprise Gateway can only identify the trunk group that the call came in on, it cannot identify the individual line on PBX A that is placing the call. The SIP URL used to identify the caller is shown in these flows as sip:IdentifierString@gw1.wcom.com. A unique IdentifierString is provisioned on the Gateway against each incoming trunk group. Note: the string could be a telephone number.

Message Details

F1 MF Digits PBX A -> GW 1

```
KP 1 972 555 2222 ST
```

F2 INVITE A -> Proxy 1

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

/* Proxy 1 uses a Location Service function to determine where the phone number +1-972-555-2222 is located. Based upon location analysis the call is forwarded to SIP User B. */

F3 INVITE Proxy 1 -> User B

```
INVITE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) Proxy 1 -> GW 1

```
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
```

F5 180 Ringing User B -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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
```

F6 180 Ringing Proxy 1 -> GW 1

```
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. */

F7 200 OK User B -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
```

```
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=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: <sip:UserB@110.111.112.113>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 147

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

F8 200 OK Proxy 1 -> GW 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=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: <sip:UserB@110.111.112.113>
Content-Type: application/sdp
Content-Length: 147

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

F9 ACK GW 1 -> Proxy 1

```
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP gw1.wcom.com:5060
Route: <sip:UserB@110.111.112.113>
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
```

F10 ACK Proxy 1 -> User B

```
ACK sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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. */

F11 BYE GW 1 -> Proxy 1

```
BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP gw1.wcom.com:5060
Route: <sip:UserB@110.111.112.113>
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
```

F12 BYE Proxy 1 -> User B

```
BYE sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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
```

F13 200 OK User B -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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


F14 200 OK Proxy 1 -> GW 1

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            GW 1          Proxy 1           User B
     |                |                |                |
     |     IAM F1     |                |                |
     |--------------->|   INVITE F2    |                |
     |                |--------------->|                |
     |                |     604 F3     |                |
     |                |<---------------|                |
     |                |     ACK F4     |                |
     |                |--------------->|                |
     |     REL F5     |                |                |
     |<---------------|                |                |
     |     RLC F6     |                |                |
     |--------------->|                |                |
     |                |                |                |
```

User A attempts to place a call through Gateway GW 1 and Proxy 1, which is unable to find any routing for the number. The call is rejected by Proxy 1 with a REL message containing a specific Cause value mapped by the gateway based on the SIP error.

Message Details

F1 IAM User A -> GW 1

```
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
```

F2 INVITE A -> Proxy 1

```
INVITE sip:+1-972-555-9999@ss1.wcom.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: <sip:+1-314-555-1111@gw1.wcom.com;user=phone>
To: <sip:+1-972-555-9999@ss1.wcom.com;user=phone>
Call-ID: 12345602@gw1.wcom.com
CSeq: 1 INVITE
Contact: <sip:+1-314-555-1111@gw1.wcom.com;user=phone>
Content-Type: application/sdp
Content-Length: 150

v=0
```

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

/* Proxy 1 uses a Location Service to find a route to +1-972-555-9999. A route is not found, so Proxy 1 rejects the call. */

F3 604 Does Not Exist Anywhere Proxy 1 -> GW 1

```
SIP/2.0 604 Does Not Exist Anywhere
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: <sip:+1-314-555-1111@gw1.wcom.com;user=phone>
To: <sip:+1-972-555-9999@ss1.wcom.com;user=phone>;tag=6a34d410
Call-ID: 12345602@gw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F4 ACK GW 1 -> Proxy 1

```
ACK sip:+1-972-555-9999@ss1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
From: <sip:+1-314-555-1111@gw1.wcom.com;user=phone>
To: <sip:+1-972-555-9999@ss1.wcom.com;user=phone>;tag=6a34d410
Call-ID: 12345602@gw1.wcom.com
CSeq: 1 ACK
Content-Length: 0
```

F5 REL GW 1 -> User A

```
REL
CauseCode=1
CodingStandard=CCITT
```

F6 RLC User A -> GW 1

```
RLC
```

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

```
   User A           NGW 1           Proxy 1          User B
     |                |                |                |
     |     IAM F1     |                |                |
     |--------------->|   INVITE F2    |                |
     |                |--------------->|   INVITE F3    |
     |                |    (100) F4    |--------------->|
     |                |<---------------|                |
     |                |                |     600 F5     |
     |                |                |<---------------|
     |                |                |     ACK F6     |
     |                |     600 F7     |--------------->|
     |                |<---------------|                |
     |                |     ACK F8     |                |
     |                |--------------->|                |
     |   REL(17) F9   |                |                |
     |<---------------|                |                |
     |     RLC F10    |                |                |
     |<-------------->|                |                |
     |                |                |                |
```

In this scenario, User A calls User B through Network Gateway NGW 1 and Proxy 1. The call is routed to User B by Proxy 1. The call is rejected by 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. For more discussion on interworking, refer to [5].

Message Details

F1 IAM User A -> NGW 1

```
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
```

F2 INVITE A -> Proxy 1

```
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@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine a route for +1-972-555-2222. The call is then forwarded to User B. */

F3 INVITE F3 Proxy 1 -> User B

```
INVITE UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) Proxy 1 -> NGW 1

```
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
```

```
   CSeq: 1 INVITE
   Content-Length: 0


   F5 600 Busy Everywhere User B -> Proxy 1

   SIP/2.0 600 Busy Everywhere
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   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@ss1.wcom.com;user=phone>;tag=314159
   Call-ID: 12345602@ngw1.wcom.com
   CSeq: 1 INVITE
   Content-Length: 0


   F6 ACK Proxy 1 -> User B

   ACK UserB@there.com SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   From: <sip:+1-314-555-1111@ngw1.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


   F7 600 Busy Everywhere Proxy 1 -> NGW 1

   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@ss1.wcom.com;user=phone>;tag=59
   Call-ID: 12345602@ngw1.wcom.com
   CSeq: 1 INVITE
   Content-Length: 0


   F8 ACK NGW 1 -> Proxy 1

   ACK UserB@there.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@ss1.wcom.com;user=phone>;tag=59
   Call-ID: 12345602@ngw1.wcom.com
   CSeq: 1 ACK
   Content-Length: 0


   F9 REL NGW 1 -> User A
```

```
REL
CauseCode=17 Busy
CodingStandard=CCITT
```

F10 RLC User A -> NGW 1

```
RLC
```

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

```
   User A           NGW 1           Proxy 1          User B
      |                 |                 |                 |
      |     IAM F1      |                 |                 |
      |---------------->|   INVITE F2     |                 |
      |                 |---------------->|   INVITE F3     |
      |                 |    (100) F4     |---------------->|
      |                 |<----------------|                 |
      |                 |                 |     600 F5      |
      |                 |                 |<----------------|
      |                 |                 |     ACK F6      |
      |                 |     600 F7      |---------------->|
      |                 |<----------------|                 |
      |                 |     ACK F8      |                 |
      |     ANM F9      |---------------->|                 |
      |<----------------|                 |                 |
      | Both Way Voice  |                 |                 |
      |<===============>|                 |                 |
      |    Busy Tone    |                 |                 |
      |<================|                 |                 |
      |   REL(16) F10   |                 |                 |
      |---------------->|                 |                 |
      |     RLC F11     |                 |                 |
      |<----------------|                 |                 |
      |                 |                 |                 |
```

In this scenario, User A calls User B through Network Gateway NGW1 and Proxy 1. The call is routed to User B by Proxy 1. The call is rejected by the User B client. NGW 1 sets up a two way voice path to User A, plays busy tone, and releases call after timeout.

NGW 1 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.

Again, note that for ETSI or ITU ISUP, a CONnect message would be sent instead of the Answer Message.

Message Details

F1 IAM User A -> NGW 1

```
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
```

F2 INVITE A -> Proxy 1

```
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@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine a route for +1-972-555-2222. The call is then forwarded to User B. */

F3 INVITE Proxy 1 -> User B

```
INVITE UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) User B -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 600 Busy Everywhere User B -> Proxy 1

```
SIP/2.0 600 Busy Everywhere
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 ACK Proxy 1 -> User B

```
ACK UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: <sip:+1-314-555-1111@ngw1.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
```

F7 600 Busy Everywhere Proxy 1 -> NGW 1

```
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@ss1.wcom.com;user=phone>;tag=66536336
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F8 ACK NGW 1 -> Proxy 1

```
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@ss1.wcom.com;user=phone>;tag=66536336
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0
```

F9 ACM NGW 1 -> User A

ANM

/* A two way speech path is established between NGW 1 and User A. */

/* Call Released after User A hangs up. */

F10 REL User A -> NGW 1

```
REL
CauseCode=16
CodingStandard=CCITT
```

F11 RLC NGW 1 -> User A

RLC

5.2.4 Unsuccessful PSTN->SIP, ACM timeout

```
   User A           NGW 1           Proxy 1          User B
     |                |                |                |
     |     IAM F1     |                |                |
     |--------------->|   INVITE F2    |                |
     |                |--------------->|   INVITE F3    |
     |                |    (100) F4    |--------------->|
     |                |<---------------|                |
     |                |                |   INVITE F5    |
     |                |                |--------------->|
     |                |                |   INVITE F6    |
     |                |                |--------------->|
     |                |                |   INVITE F7    |
     |                |                |--------------->|
     |                |                |   INVITE F8    |
     |                |                |--------------->|
     |                |                |   INVITE F9    |
     |                |                |--------------->|
     |     REL F10    |                |                |
     |--------------->|                |                |
     |     RLC F11    |                |                |
     |<---------------|                |                |
     |                |   CANCEL F12   |                |
     |                |--------------->|                |
     |                |    200 F13     |                |
     |                |<---------------|   CANCEL F14   |
     |                |                |--------------->|
     |                |                |   CANCEL F15   |
     |                |                |--------------->|
     |                |                |   CANCEL F16   |
     |                |                |--------------->|
     |                |                |   CANCEL F17   |
     |                |                |--------------->|
     |                |                |   CANCEL F18   |
     |                |                |--------------->|
     |                |                |                |
```

User A calls User B through NGW 1 and Proxy 1. Proxy 1 re-sends the INVITE after the expiration of SIP timer T1 without receiving any response from User B. User B never responds with 180 Ringing or any other response (it is reachable but unresponsive). After the expiration of a timer, User A’s network disconnects the call by sending a Release message REL. The Gateway maps this to a CANCEL which is again re-sent by Proxy 1 after SIP T1 timer expires.

Message Details

F1 IAM User A -> NGW 1

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
```

F2 INVITE A -> Proxy 1

```
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@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine a route for +1-972-555-2222. The call is then forwarded to User B. */

F3 INVITE Proxy 1 -> User B

```
INVITE sip:UserB@there.com  SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.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
c c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 100 Trying Proxy 1 -> NGW 1

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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0

F5 INVITE Proxy 1 -> User B

Same as Message F3

F6 INVITE Proxy 1 -> User B

Same as Message F3

F7 INVITE Proxy 1 -> User B

Same as Message F3

F8 INVITE Proxy 1 -> User B

Same as Message F3

F9 INVITE Proxy 1 -> User B

Same as Message F3

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

F10 REL User A -> NGW 1

REL
CauseCode=16 Normal
CodingStandard=CCITT

F11 RLC NGW 1 -> User A

RLC

F12 CANCEL NGW 1 -> Proxy 1

```
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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0
```

F13 200 OK Proxy 1 -> NGW 1

```
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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0
```

F14 CANCEL Proxy 1 -> User B

```
CANCEL sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0
```

F15 CANCEL Proxy 1 -> User B

Same as Message F14

F16 CANCEL Proxy 1 -> User B

Same as Message F14

F17 CANCEL Proxy 1 -> User B

Same as Message F14

F18 CANCEL Proxy 1 -> User B

Same as Message F14

F19 CANCEL Proxy 1 -> User B

Same as Message F14

5.2.5 Unsuccessful PSTN->SIP, ACM timeout, stateless Proxy

```
User A           NGW 1      Stateless Proxy 1     User B
   |                 |                 |                 |
   |      IAM F1     |                 |                 |
   |---------------->|   INVITE F2     |                 |
   |                 |---------------->|   INVITE F3     |
   |                 |   INVITE F4     |---------------->|
   |                 |---------------->|   INVITE F5     |
   |                 |   INVITE F6     |---------------->|
   |                 |---------------->|   INVITE F7     |
   |                 |   INVITE F8     |---------------->|
   |                 |---------------->|   INVITE F9     |
   |                 |   INVITE F10    |---------------->|
   |                 |---------------->|   INVITE F11    |
   |                 |   INVITE F12    |---------------->|
   |                 |---------------->|   INVITE F13    |
   |                 |                 |---------------->|
   |      REL F14    |                 |                 |
   |---------------->|                 |                 |
   |      RLC F15    |                 |                 |
   |<----------------|                 |                 |
   |                 |   CANCEL F16    |                 |
   |                 |---------------->|   CANCEL F17    |
   |                 |   CANCEL F18    |---------------->|
   |                 |---------------->|   CANCEL F19    |
   |                 |   CANCEL F20    |---------------->|
   |                 |---------------->|   CANCEL F21    |
   |                 |   CANCEL F22    |---------------->|
   |                 |---------------->|   CANCEL F23    |
   |                 |   CANCEL F24    |---------------->|
   |                 |---------------->|   CANCEL F25    |
   |                 |   CANCEL F26    |---------------->|
   |                 |---------------->|   CANCEL F27    |
   |                 |                 |---------------->|
   |                 |                 |                 |
```

In this scenario, User A calls User B through NGW 1 and Proxy 1. Since Proxy 1 is stateless (it does not send a 100 Trying response), NGW 1 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 (CauseCode=102 Timeout).

Message Details

F1 IAM User A -> NGW 1

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


   F2 INVITE NGW 1 -> Proxy 1

   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@ss1.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
   s=Session SDP
   c=IN IP4 gatewayone.wcom.com
   t=0 0
   m=audio 3456 RTP/AVP 0
   a=rtpmap:0 PCMU/8000


   /* Proxy 1 uses a Location Service function to determine a route for
   +1-972-555-2222.  The call is then forwarded to User B. */

   F3 INVITE Proxy 1 -> User B

   INVITE sip:UserB@there.com  SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   Via: SIP/2.0/UDP ngw1.wcom.com:5060
   Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
   From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
   To: <sip:+1-972-555-2222@ss1.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
   s=Session SDP
   c=IN IP4 gatewayone.wcom.com
   t=0 0
```

```
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 INVITE NGW 1 -> Proxy 1

Same as Message F2

F5 INVITE Proxy 1 -> User B

Same as Message F3

F6 INVITE NGW 1 -> Proxy 1

Same as Message F2

F7 INVITE Proxy 1 -> User B

Same as Message F3

F8 INVITE NGW 1 -> Proxy 1

Same as Message F2

F9 INVITE Proxy 1 -> User B

Same as Message F3

F10 INVITE NGW 1 -> Proxy 1

Same as Message F2

F11 INVITE Proxy 1 -> User B

Same as Message F3

F12 INVITE NGW 1 -> Proxy 1

Same as Message F2

F13 INVITE Proxy 1 -> User B

Same as Message F3

/* A timer expires in User A’s access network. */

F14 REL User A -> NGW 1

```
REL
CauseCode=102 Timeout
CodingStandard=CCITT
```

F15 RLC NGW 1 -> User A

```
RLC
```

F16 CANCEL NGW 1 -> Proxy 1

```
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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0
```

F17 CANCEL Proxy 1 -> User B

```
CANCEL sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0
```

F18 CANCEL NGW 1 -> Proxy 1

Same as Message F16

F19 CANCEL Proxy 1 -> User B

Same as Message F17

F20 CANCEL NGW 1 -> Proxy 1

Same as Message F16

F21 CANCEL Proxy 1 -> User B

Same as Message F17

F22 CANCEL NGW 1 -> Proxy 1

Same as Message F16

F23 CANCEL Proxy 1 -> User B

Same as Message F17

F24 CANCEL NGW 1 -> Proxy 1

Same as Message F16

F25 CANCEL Proxy 1 -> User B

Same as Message F17

F26 CANCEL NGW 1 -> Proxy 1

Same as Message F16

F27 CANCEL Proxy 1 -> User B

Same as Message F17

5.2.6 Unsuccessful PSTN->SIP, ANM timeout

```
   User A           NGW 1          Proxy 1          User B
     |                |                |                |
     |     IAM F1     |                |                |
     |--------------->|   INVITE F2    |                |
     |                |--------------->|   INVITE F3    |
     |                |    (100) F4    |--------------->|
     |                |<---------------|                |
     |                |                |     180 F5     |
     |                |    180 F6      |<---------------|
     |     ACM F7     |<---------------|                |
     |<---------------|                |                |
     |  One Way Voice |                |                |
     |<===============|                |                |
     |  Ringing Tone  |                |                |
     |<===============|                |                |
     |                |                |                |
     |     REL F8     |                |                |
     |--------------->|                |                |
     |     RLC F9     |                |                |
     |<---------------|   CANCEL F10   |                |
     |                |--------------->|                |
     |                |     200 F11    |                |
     |                |<---------------|                |
     |                |                |   CANCEL F12   |
     |                |                |--------------->|
     |                |                |    200 F13     |
     |                |                |<---------------|
     |                |                |    487 F14     |
     |                |                |<---------------|
     |                |                |    ACK F15     |
     |                |     487 F16    |--------------->|
     |                |<---------------|                |
     |                |     ACK F17    |                |
     |                |--------------->|                |
     |                |                |                |
```

In this scenario, User A calls User B through NGW 1 and Proxy 1. User B does not respond with 200 OK. NGW 1 plays ringing tone since the ACM indicates that interworking has been encountered. User A disconnects the call with a Release message REL which is mapped by NGW 1 to a CANCEL. Note that if User B had sent a 200 OK response after the REL, NGW 1 would have sent an ACK then a BYE to properly terminate the call.

Message Details

F1 IAM User A -> NGW 1

```
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
```

F2 INVITE A -> Proxy 1

```
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@ss1.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
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine a route for +1-972-555-2222. The call is then forwarded to User B. */

F3 INVITE Proxy 1 -> User B

```
INVITE sip:UserB@there.com  SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.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
```

```
s=Session SDP
c=IN IP4 gatewayone.wcom.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 (100 Trying) User B -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 180 Ringing User B -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 180 Ringing Proxy 1 -> NGW 1

```
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F7 ACM NGW 1 -> 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 A hangs up */

F8 REL User A -> NGW 1

REL
CauseCode=16 Normal
CodingStandard=CCITT


F9 RLC NGW 1 -> User A

RLC


F10 CANCEL NGW 1 -> Proxy 1

CANCEL sip:UserB@there.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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0


F11 200 OK Proxy 1 -> NGW 1

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@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0


F12 CANCEL Proxy 1 -> User B

CANCEL sip:UserB@110.111.112.113 SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0
```

F13 200 OK User B -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.wcom.com;user=phone>
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 CANCEL
Content-Length: 0
```

F14 487 Request Cancelled User B -> Proxy 1

```
SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.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 INVITE
Content-Length: 0
```

F15 ACK Proxy 1 -> User B

```
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@ss1.wcom.com;user=phone>;tag=314159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 ACK
Content-Length: 0
```

F16 487 Request Cancelled Proxy 1 -> NGW 1

```
SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-972-555-2222@ss1.wcom.com;maddr=ss1.wcom.com>
From: <sip:+1-314-555-1111@ngw1.wcom.com;user=phone>
To: <sip:+1-972-555-2222@ss1.wcom.com;user=phone>;tag=14159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F17 ACK NGW 1 -> Proxy 1

```
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@ss1.wcom.com;user=phone>;tag=14159
Call-ID: 12345602@ngw1.wcom.com
CSeq: 1 ACK
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.

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 [5].

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 indicate in-band alerting may flow from the called party telephone switch to the caller.

6.1.1 Successful ISUP PSTN to ISUP PSTN call

```
     User A         NGW 1          Proxy 1        GW 2           User C
    |              |              |              |              |
    |     IAM F1   |              |              |              |
    |------------->|              |              |              |
    |              |  INVITE F2   |              |              |
    |              |------------->|  INVITE F3   |              |
    |              |              |------------->|     IAM F4   |
    |              |              |              |------------->|
    |              |              |              |     ACM F5   |
    |              |              |   183 F6     |<-------------|
    |              |    183 F7    |<-------------|              |
    |    ACM F8    |<-------------|              |              |
    |<-------------|              |              |              |
    | One Way Voice|      One Way RTP Media      | One Way Voice|
    |<=============|<============================|<=============|
    |              |              |              |    ANM F9    |
    |              |              |   200 F10    |<-------------|
    |              |    200 F11   |<-------------|              |
    |    ANM F12   |<-------------|              |              |
    |<-------------|              |              |              |
    |              |    ACK F13   |              |              |
    |              |------------->|    ACK F14   |              |
    |              |              |------------->|              |
    |Both Way Voice|     Both Way RTP Media      |Both Way Voice|
    |<=============|<===========================>|<=============|
    |              |              |              |    REL F15   |
    |              |              |              |<-------------|
    |              |              |   BYE F16    |              |
    |              |    BYE F18   |<-------------|    RLC F17   |
    |              |<-------------|              |------------->|
    |              |              |              |              |
    |              |    200 F19   |              |              |
    |              |------------->|    200 F20   |              |
    |              |              |------------->|              |
    |    REL F21   |              |              |              |
    |<-------------|              |              |              |
    |    RLC F22   |              |              |              |
    |------------->|              |              |              |
    |              |              |              |              |
```

In this scenario, User A in the PSTN calls User C who is an extension on a PBX. User A’s telephone switch signals via SS7 to the Network Gateway NGW 1, while User C’s PBX signals via SS7 with the Enterprise Gateway GW 2. The CdPN and CgPN are mapped by GW1 into SIP URLs and placed in the To and From headers. Proxy 1 looks up the dialed digits in the Request-URI and maps the digits to the PBX extension of User C served by GW 2. The Request-URI in F3 uses the host portion of the Request-URI to identify what private dialing plan is being referenced. The INVITE is then forwarded to GW 2 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

F1 IAM User A -> NGW 1

```
IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=918-555-3333,NPI=E.164,NOA=National
USI=Speech
CPT=0 0
C=Normal
CCI=Not Required
```

F2 INVITE NGW 1 -> Proxy 1

```
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: 149

v=0
o=GW1 2890844526 2890844526 IN IP4 gw1.wcom.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 consults Location Service and translates the dialed number to a private number in the Request-URI*/

F3 INVITE Proxy 1 -> GW 2

```
INVITE sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
```

```
   Via: SIP/2.0/UDP gw1.wcom.com:5060
   Record-Route: <sip:+1-918-555-3333@ss1.wcom.com;maddr=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: 149

   v=0
   o=GW1 2890844526 2890844526 IN IP4 gw1.wcom.com
   s=Session SDP
   c=IN IP4 100.101.102.103
   t=0 0
   m=audio 49172 RTP/AVP 0
   a=rtpmap:0 PCMU/8000


   F4 IAM GW 2 -> 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


   F5 ACM User C -> GW 2

   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, GW 2 returns a 183 response.  In-band
   call progress indications are sent to User A through NGW 1. */

   F6 183 Session Progress GW 2 -> Proxy 1

   SIP/2.0 183 Session Progress
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
```

```
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-Length: 0
```

F7 183 Session Progress Proxy 1 -> GW 1

```
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-Length: 0
```

/* NGW 1 receives packets from GW 2 with encoded ringback, tones or other audio. NGW 1 decodes this and places it on the originating trunk. */

F8 ACM NGW 1 -> 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 */

F9 ANM User C -> GW 2

```
ANM
```

F10 200 OK GW 2 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-918-555-3333@ss1.wcom.com;maddr=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@wcom.com>
Content-Type: application/sdp
Content-Length: 149

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

F11 200 OK Proxy 1 -> NGW 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Record-Route: <sip:+1-918-555-3333@ss1.wcom.com;maddr=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@wcom.com>
Content-Type: application/sdp
Content-Length: 149

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

F12 ANM NGW 1 -> User A

ANM

F13 ACK NGW 1 -> Proxy 1

```
ACK sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP ngw1.wcom.com:5060
Route: <sip:444-3333@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@gw1.wcom.com
```

```
   CSeq: 1 ACK
   Content-Length: 0


   F14 ACK Proxy 1 -> GW 2

   ACK sip:444-3333@wcom.com SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   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 NGW 1 and GW 2. */

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

   F15 REL User C -> GW 2

   REL
   CauseCode=16 Normal
   CodingStandard=CCITT


   F16 BYE GW 2 -> Proxy 1

   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


   F17 RLC GW 2 -> User C

   RLC


   F18 BYE Proxy 1 -> NGW 1

   BYE sip:+1-314-555-1111@gw1.wcom.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   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;user=phone>
```

```
   Call-ID: 12345600@ngw1.wcom.com
   CSeq: 4 BYE
   Content-Length: 0


   F19 200 OK NGW 1 -> Proxy 1

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
   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;user=phone>
   Call-ID: 12345600@ngw1.wcom.com
   CSeq: 4 BYE
   Content-Length: 0


   F20 200 OK Proxy 1 -> GW 2

   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;user=phone>
   Call-ID: 12345600@ngw1.wcom.com
   CSeq: 4 BYE
   Content-Length: 0


   F21 REL User C -> GW 2

   REL
   CauseCode=16 Normal
   CodingStandard=CCITT


   F22 RLC GW 2 -> User C

   RLC
```

6.1.2 Successful FGB PBX to ISDN PBX call with overflow

```
   User A       GW 1        Proxy 1      GW 2         GW 3        User B
   |            |            |            |            |            |
   |  Seizure   |            |            |            |            |
   |----------->|            |            |            |            |
   |    Wink    |            |            |            |            |
   |<-----------|            |            |            |            |
   |MF Digits F1|            |            |            |            |
   |----------->|            |            |            |            |
   |            | INVITE F2  |            |            |            |
   |            |----------->| INVITE F3  |            |            |
   |            |            |----------->|            |            |
   |            |            |   503 F4   |            |            |
   |            |            |<-----------|            |            |
   |            |            |   ACK F5   |            |            |
   |            |            |----------->|            |            |
   |            |            |  INVITE F6              |            |
   |            |            |------------------------>|  SETUP F7  |
   |            |            |         (100) F8        |----------->|
   |            |            |<------------------------|CALL PROC F9|
   |            |            |                         |<-----------|
   |            |            |                         | ALERT F10  |
   |            |            |          183 F11        |<-----------|
   |            |  183 F12   |<------------------------|            |
   |            |<-----------|                         |            |
   |OneWay Voice|           One Way RTP Media          |OneWay Voice|
   |<===========|<=====================================|<===========|
   |            |            |                         | CONNect F13|
   |            |            |         200 F14         |<-----------|
   |            |  200 F15   |<------------------------|            |
   |  Seizure   |<-----------|                         |            |
   |<-----------|  ACK F16   |                         |            |
   |            |----------->|         ACK F17         |            |
   |            |            |------------------------>|CONN ACK F18|
   |            |            |                         |----------->|
   |BothWayVoice|          Both Way RTP Media          |BothWayVoice|
   |<==========>|<====================================>|<==========>|
   |            |            |                         |  DISC F19  |
   |            |            |                         |<-----------|
   |            |            |         BYE F20         |            |
   |            |  BYE F21   |<------------------------|  REL F22   |
   |Seiz Removal|<-----------|                         |----------->|
   |<-----------|  200 F23   |                         |            |
   |Seiz Removal|----------->|         200 F24         |            |
   |----------->|            |------------------------>| REL COM F25|
   |            |            |                         |<-----------|
   |            |            |                         |            |
```

PBX User A calls PBX User C via Gateway GW 1 and Proxy 1. During the attempt to reach User C via GW 2, an error is encountered - Proxy 1 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 GW 2. Proxy 1 recognizes the error and uses an alternative route via GW 3 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 -> GW 1

Seizure

GW 1 -> PBX A

Wink

F1 MF Digits PBX A -> GW 1

KP 444 3333 ST

F2 INVITE GW 1 -> Proxy 1

```
INVITE sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: <sip:444-3333@wcom.com>
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
o=PBX_A 2890844526 2890844526 IN IP4 gw1.wcom.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine where B is located. Response is returned listing alternative routes, GW2 and GW3, which are then tried sequentially. */

F3 INVITE Proxy 1 -> GW 2

```
INVITE sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:444-3333@wcom.com;maddr=ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: <sip:444-3333@wcom.com>
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
o=PBX_A 2890844526 2890844526 IN IP4 gw1.wcom.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F4 503 Service Unavailable GW 2 -> Proxy 1

```
SIP/2.0 503 Service Unavailable
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: <sip:444-3333@wcom.com>;tag=314159
Call-ID: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 ACK Proxy 1 -> GW 2

```
ACK sip:444-3333@wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.1
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: <sip:444-3333@wcom.com>;tag=314159
Call-ID: 12345600@gw1.wcom.com
CSeq: 1 ACK
Content-Length: 0
```

F6 INVITE Proxy 1 -> GW 3

```
INVITE sip:+1-918-555-3333@gw3.wcom.com;user=phone SIP/2.0
```

```
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.2
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:444-3333@wcom.com;maddr=ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: <sip:444-3333@wcom.com>
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
o=PBX_A 2890844526 2890844526 IN IP4 gw1.wcom.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F7 SETUP GW 3 -> PBX C

Protocol discriminator=Q.931
Call reference: Flag=0
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

F8 (100 Trying) GW 3 -> Proxy 1

```
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@wcom.com>
Call-ID: 12345600@gw1.wcom.com
CSeq: 1 INVITE
Content-Length: 0
```

F9 CALL PROCeeding PBX C -> GW 3

Protocol discriminator=Q.931
Call reference: Flag=1

```
   Message type=CALL PROC
   Channel identification=Exclusive B-channel


   F10 ALERT PBX C -> GW 3

   Protocol discriminator=Q.931
   Call reference: Flag=1
   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, GW 3 returns a 183 response.  In-band
   call progress indications are then sent to the originator.  */

   F11 183 Session Progress GW 3 -> Proxy 1

   SIP/2.0 183 Session Progress
   Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.2
   Via: SIP/2.0/UDP gw1.wcom.com:5060
   From: PBX_A <sip:IdentifierString@gw1.wcom.com>
   To: <sip:444-3333@wcom.com>;tag=123456789
   Call-ID: 12345600@gw1.wcom.com
   CSeq: 1 INVITE
   Content-Length: 0


   F12 183 Session Progress Proxy 1 -> GW 1

   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@wcom.com>;tag=123456789
   Call-ID: 12345600@gw1.wcom.com
   CSeq: 1 INVITE
   Content-Length: 0


   /* GW 1 receives packets from GW 3 with encoded ringback, tones or
   other audio.  GW 1 decodes this and places it on the originating
   trunk. */

   F13 CONNect PBX C -> GW 3

   Protocol discriminator=Q.931
   Call reference: Flag=1
   Message type=CONN


   F14 200 OK GW 3 -> Proxy 1
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.2
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:444-3333@wcom.com;maddr=ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: <sip:444-3333@wcom.com>;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: 149

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

F15 200 OK Proxy 1 -> GW 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP gw1.wcom.com:5060
Record-Route: <sip:444-3333@wcom.com;maddr=ss1.wcom.com>
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: <sip:444-3333@wcom.com>;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: 149

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

GW 1 -> PBX A

Seizure

F16 ACK GW 1 -> Proxy 1

```
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:444-3333@wcom.com>;tag=123456789
Call-ID: 12345600@gw1.wcom.com
CSeq: 1 ACK
Content-Length: 0
```

F17 ACK Proxy 1 -> GW 3

```
ACK sip:+1-918-555-3333@gw3.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.2
Via: SIP/2.0/UDP gw1.wcom.com:5060
From: PBX_A <sip:IdentifierString@gw1.wcom.com>
To: <sip:444-3333@wcom.com>;tag=123456789
Call-ID: 12345600@gw1.wcom.com
CSeq: 1 ACK
Content-Length: 0
```

F18 CONNect ACK GW 3 -> PBX C

```
Protocol discriminator=Q.931
Call reference: Flag=0
Message type=CONN ACK
```

/* RTP streams are established between GW 1 and GW 3. */

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

F19 DISConnect PBX C -> GW 3

```
Protocol discriminator=Q.931
Call reference: Flag=1
Message type=DISC
Cause=16 (Normal clearing)
```

F20 BYE GW 3 -> Proxy 1

```
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@wcom.com>;tag=123456789
To: PBX_A <sip:IdentifierString@gw1.wcom.com>
Call-ID: 12345600@gw1.wcom.com
CSeq: 1 BYE
Content-Length: 0
```

F21 BYE Proxy 1 -> GW 1

```
BYE sip:IdentifierString@gw1.wcom.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.2
Via: SIP/2.0/UDP gw3.wcom.com:5060
From: <sip:444-3333@wcom.com>;tag=123456789
To: PBX_A <sip:IdentifierString@gw1.wcom.com>
Call-ID: 12345600@gw1.wcom.com
CSeq: 1 BYE
Content-Length: 0
```

GW 1 -> PBX A

Seizure removal

F22 RELease GW 3 -> PBX C

```
Protocol discriminator=Q.931
Call reference: Flag=0
Message type=REL
```

F23 200 OK GW 1 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=2d4790.2
Via: SIP/2.0/UDP gw3.wcom.com:5060
From: <sip:444-3333@wcom.com>;tag=123456789
To: PBX_A <sip:IdentifierString@gw1.wcom.com>
Call-ID: 12345600@gw1.wcom.com
CSeq: 1 BYE
Content-Length: 0
```

F24 200 OK Proxy 1 -> GW 3

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

F25 RELease COMplete PBX C -> GW 3

Protocol discriminator=Q.931
Call reference: Flag=1
Message type=REL COM

PBX A -> GW 1

Seizure removal

7 SIP Test Messages

The files in here are test messages for SIP servers to exercise various functions. They have been used in SIP "bakeoff" interoperablity events. All messages shown here are valid, unless otherwise noted. The correct behavior of servers and clients is also described.

7.1 INVITE Parser Torture Test Message

This message is a correctly formatting SIP message. It contains:

line folding all over
escaped characters within quotes
LWS between colons, semicolons, headers, and other fields
both comma separated and separate listing of headers
mix or short and long form for the same header
unknown header field
unusual header ordering
nested comments
unknown parameters of a known header

Proxies should forward message and clients should respond as to a normal INVITE message.

Message Details

```
INVITE sip:vivekg@chair.dnrc.bell-labs.com SIP/2.0
TO :
 sip:vivekg@chair.dnrc.bell-labs.com ;   tag    = 1918181833n
From   : "J Rosenberg \\\"" <sip:jdrosen@lucent.com>
  ;
  tag = 98asjd8
Call-ID
 : 0ha0isndaksdj@10.1.1.1
cseq: 8
  INVITE
Via  : SIP  /   2.0
 /UDP
    135.180.130.133
Subject :
NewFangledHeader:   newfangled value
 more newfangled value
Content-Type: application/sdp
v:  SIP  / 2.0  / TCP     12.3.4.5   ;
  branch  =   9ikj8  ,
 SIP  /    2.0   / UDP  1.2.3.4   ; hidden
m:"Quoted string \"\"" <sip:jdrosen@bell-labs.com> ; newparam =
newvalue ;
  secondparam = secondvalue  ; q = 0.33
```

```
     (((nested comments) and (more)))   ,
    tel:4443322

   v=0
   o=mhandley 29739 7272939 IN IP4 126.5.4.3
   c=IN IP4 135.180.130.88
   m=audio 492170 RTP/AVP 0 12
   m=video 3227 RTP/AVP 31
   a=rtpmap:31 LPC
```

7.2 INVITE with Proxy-Require and Require

This message tests support for Proxy-Require and Require. It is a request that contains both headers, listing new features.

Proxies and clients should respond with a 420 Bad Extension, and an Unsupported header listing these features.

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: sip:j_user@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.1.1.1
Require: newfeature1, newfeature2
Proxy-Require: newfeature3, newfeature4
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
```

7.3 INVITE with Unknown Schemes in URIs and URLs

This message contains unknown schemes in the Request URI, To, From and Contact headers of a request.

A server should probably return a not found error; but other behaviors are acceptable.

Message Details

```
INVITE name:John_Smith SIP/2.0
To: isbn:2983792873
From: http://www.cs.columbia.edu
Call-ID: 0ha0isndaksdj@10.1.2.3
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp

v=0
```

```
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.4 REGISTER with Y2038 Test

This message is a registration request with an expiration year of 2040. This makes sure that a server doesn’t crash on seeing a date past Y2038.

The correct behavior is probably to limit the lifetime to some configured maximum.

Message Details

```
REGISTER sip:company.com SIP/2.0
To: sip:user@company.com
From: sip:user@company.com
Contact: sip:user@host.company.com
Call-ID: 0ha0isndaksdj@10.0.0.1
CSeq: 8 REGISTER
Via: SIP/2.0/UDP 135.180.130.133
Expires: Sat, 01 Dec 2040 16:00:00 GMT
```

7.5 INVITE with inconsistent Accept and message body

This is a UAS test. It is a request that includes an Accept header without SDP. The UAS should respond with an error.

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: sip:j_user@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.0.0.1
Accept: text/newformat
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp

v=0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
```

```
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.6 INVITE with non-SDP message body

This is a test of a user agent server. It is a request that includes a body of a non-SDP type.

The user agent server should respond with an error.

Message Details

```
INVITE sip:user@comapny.com SIP/2.0
To: sip:j.user@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.0.0.1
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/newformat

<audio>
 <pcmu port="443"/>
</audio>
```

7.7 Unknown Method Message

This request message contains a new unknown method, NEWMETHOD.

A proxy should forward this using the same retransmission rules as BYE. A UAS should reject it with an error, and list the available methods in the response.

Message Details

```
NEWMETHOD sip:user@comapny.com SIP/2.0
To: sip:j.user@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.0.0.1
CSeq: 8 NEWMETHOD
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
```

```
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.8 Unknown Method with CSeq Error

This message is nearly identical to the Unknown Method message. It is a request with a new unknown method, but with a CSeq method tag which does not match.

A proxy should either respond with an error, or correct the method tag. The user agent should reject it with an error, and list the available methods in the response.

Message Details

```
NEWMETHOD sip:user@comapny.com SIP/2.0
To: sip:j.user@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.0.1.1
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.9 REGISTER with Unknown Authorization Scheme

This message is a REGISTER request with an unknown authorization scheme.

The server should do something reasonable, such as rejecting the request.

Message Details

```
REGISTER sip:company.com SIP/2.0
To: sip:j.user@company.com
From: sip:j.user@company.com
Call-ID: 0ha0isndaksdj@10.0.1.1
```

```
CSeq: 8 REGISTER
Via: SIP/2.0/UDP 135.180.130.133
Authorization: Super-PGP ajsohdaosdh0asyhdaind08yasdknasd09asidhas0d8
```

7.10 Multiple SIP Request in a Single Message

This message contains two requests, separated by a bunch of whitespace. Since the message exceeds the length indicated in the Content-Length header, the message should be rejected. (Multiple SIP requests per UDP packet are no longer allowed.)

Message Details

```
REGISTER sip:company.com SIP/2.0
To: sip:j.user@company.com
From: sip:j.user@company.com
Call-ID: 0ha0isndaksdj@10.0.2.2
Contact: sip:j.user@host.company.com
CSeq: 8 REGISTER
Via: SIP/2.0/UDP 135.180.130.133
Content-Length: 0



INVITE sip:joe@company.com SIP/2.0
To: sip:joe@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isnda977644900765@10.0.0.1
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m =video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.11 INVITE missing Required Headers

This message contains no Call-ID, From, or To header.

The server should not crash, and ideally should respond with an error.

Message Details

```
INVITE sip:user@company.com SIP/2.0
CSeq: 0 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.12 INVITE with Duplicate Required Headers

The message contains a request with an extra Call-ID and To field.

The server should not crash, and should ideally respond with an error.

Message Details

```
INVITE sip:user@company.com SIP/2.0
Via: SIP/2.0/UDP 135.180.130.133
CSeq: 0 INVITE
Call-ID: 98asdh@10.1.1.1
Call-ID: 98asdh@10.1.1.2
From: sip:caller@university.edu
From: sip:caller@organization.org
To: sip:user@company.com
Content-Type: application/sdp

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.13 INVITE with Illegal Expires Header

This message contains an Expires header which has illegal values for a number of components, but otherwise is syntactically correct.

Message Details

```
INVITE sip:user@company.com SIP/2.0
Via: SIP/2.0/UDP 135.180.130.133
CSeq: 0 INVITE
Call-ID: 98asdh@10.1.1.2
Expires: Thu, 44 Dec 19999 16:00:00 EDT
From: sip:caller@university.edu
To: sip:user@company.com
Content-Type: application/sdp

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.14 200 OK Response with Broadcast Via Header

This message is a response with a 2nd Via header of 255.255.255.255.

On receiving this response, the top Via header is stripped and the packet forwarded. Since the next address is the broadcast address, it causes the packet to be broadcast onto the network. A smart server should ignore packets with 2nd Via headers that are 255.255.255.255 or 127.0.0.1. At the very least it should not crash.

Message Details

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 135.180.130.57;branch=0
Via: SIP/2.0/UDP 255.255.255.255;branch=0
Call-ID: 0384840201@10.1.1.1
CSeq: 0 INVITE
From: sip:user@company.com
To: sip:user@university.edu;tag=2229
Content-Type: application/sdp

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 224.2.17.12/127
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.15 INVITE with Invalid Via and Contact Headers

This is a request with the Via and Contact headers incorrect. They contain additional semicolons and commas without parameters or values.

The server should respond with a Bad Request error.

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: sip:j.user@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.0.0.1
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133;;,;
Contact: "" <> ;,"Joe" <sip:joe@org.org>;;,,;;
Content-Type: application/sdp

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.16 INVITE with Incorrect Content-Length Header

This is a request message with a Content Length that is much larger than the length of the body.

When sent UDP, the server should respond with an error. With TCP, there’s not much you can do but wait...

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: sip:j.user@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.0.0.1
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: 9999
```

```
v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.17 INVITE with Invalid Value for Content-Length

This is a request message with a negative value for Content-Length.

The server should respond with an error.

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: sip:j.user@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.0.0.1
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: -999

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.18 INVITE with Garbage after Message Body

This is a request message with garbage after the end of the SDP included in the body.

The servers should reject the request as the body is longer than the Content-Length.

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: sip:j.user@company.com
```

```
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.0.0.1
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: 138

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
asdpasd08asdsdk:;;asd
 a0sdjhg8a0''...'';;;;
```

7.19 INVITE with Error in Display Name in To Header

This is a request with an unterminated quote in the display name of the To field.

The server can either return an error, or proxy it if it is successful parsing without the terminating quote.

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: "Mr. J. User <sip:j.user@company.com>
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.0.0.1
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: 138

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.20 INVITE with a Semicolon-Separated Parameter in the "user" Part

This is an INVITE request with a semicolon-separated parameter in the "user" part.

Outbound proxies should direct it appropriately.

Message Details

```
INVITE sip:user;par=u%40h.com@company.com SIP/2.0
To: sip:j_user@company.com
From: sip:caller@university.edu
Call-ID: 0ha0isndaksdj@10.1.1.1
CSeq: 8 INVITE
Via: SIP/2.0/UDP 135.180.130.133
```

7.21 INVITE with Illegal Enclosing of Request-URI in "<>"

This INVITE is illegal because the Request-URI has been enclosed within in "<>".

An intelligent server may be able to deal with this and fix up the Request-URI if acting as a Proxy. If not it should respond 400 with an appropriate reason phrase.

Message Details

```
INVITE <sip:user@company.com> SIP/2.0
To: sip:user@company.com
From: sip:caller@university.edu
Call-ID: 1@10.0.0.1
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: 174

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.22 INVITE with Illegal LWS within Elements of Request-URI

This INVITE has illegal LWS within the SIP URL.

An intelligent server may be able to deal with this and fix up the Request-URI if acting as a Proxy. If not it should respond 400 with an appropriate reason phrase.

Message Details

```
INVITE sip:user@company.com; transport=udp SIP/2.0
To: sip:user@company.com
From: sip:caller@university.edu
Call-ID: 2@10.0.0.1
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: 174

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.23 INVITE with illegal >1 SP between elements of Request URI

This INVITE has illegal >1 SP between elements of the Request-URI.

An intelligent server may be able to deal with this and fix up the Request-URI if acting as a Proxy. If not it should respond 400 with an appropriate reason phrase.

Message Details

```
INVITE sip:user@company.com  SIP/2.0
To: sip:user@company.com
From: sip:caller@university.edu
Call-ID: 3@10.0.0.1
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: 174

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
```

```
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.24 INVITE with a legal SIP URL containing escaped characters

This INVITE is legal and has a Request-URI with a SIP URL containing escaped characters.

Message Details

```
INVITE sip:sip%3Auser%40example.com@company.com;other-param=summit
SIP/2.0
To: sip:user@company.com
From: sip:caller@university.edu
Call-ID: 4@10.0.0.1
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: 174

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.25 INVITE with the illegal use of escaped headers in Request-URI

This INVITE is illegal as it the Request-URI contains a SIP URL containing
escaped headers.

An intelligent server may be liberal enough to accept this. A server acting as a proxy should remove the escaped header before processing.

Message Details

```
INVITE sip:user@company.com?Route=%3Csip:sip.example.com%3E SIP/2.0
To: sip:user@company.com
From: sip:caller@university.edu
```

```
Call-ID: 5@10.0.0.1
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: 174

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.26 INVITE containing an unknown in the Request URI

This INVITE contains an unknown URI scheme in the Request-URI.

A server should reject this message with a 400 response plus an appropriate reason phrase despite being able to understand the To header as a SIP URL.

Message Details

```
INVITE name:user SIP/2.0
To: sip:user@company.com
From: sip:caller@university.edu
Call-ID: 6@10.0.0.1
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Content-Type: application/sdp
Content-Length: 174

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.27 OPTIONS with no LWS between display name and <

This OPTIONS request is legal despite there being no LWS between the display name and < in the From header.

Message Details

```
OPTIONS sip:user@company.com SIP/2.0
To: sip:user@company.com
From: "caller"<sip:caller@example.com>
Call-ID: 1234abcd@10.0.0.1
CSeq: 1 OPTIONS
Via: SIP/2.0/UDP 135.180.130.133
```

7.28 OPTIONS with extran LWS between display name and <

This OPTIONS request is legal despite there being extra LWS between the display name and < in the From header.

Message Details

```
OPTIONS sip:user@company.com SIP/2.0
To: sip:user@company.com
From: "caller"    <sip:caller@example.com>
Call-ID: 1234abcd@10.0.0.1
CSeq: 2 OPTIONS
Via: SIP/2.0/UDP 135.180.130.133
```

7.29 INVITE with an illegal SIP Date format.

This INVITE is illegal as it contains a non GMT time zone in the SIP Date of the Expires header.

An intelligent server may be able to fix this up and correct the time to GMT. Alternatively this message may illicit a 400 response with an appropriate reason phrase.

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: sip:user@company.com
From: sip:caller@university.edu
Call-ID: 7@10.0.0.1
CSeq: 1 INVITE
```

```
Via: SIP/2.0/UDP 135.180.130.133
Expires: Fri, 01 Jan 2010 16:00:00 EST
Content-Type: application/sdp
Content-Length: 174

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.30 INVITE with Passed Expries Time

This is a legal INVITE but the message content has long since expired.

A server should respond 408 (Timeout).

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: sip:user@company.com
From: sip:caller@university.edu
Call-ID: 8@10.0.0.1
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Expires: Thu, 01 Dec 1994 16:00:00 GMT
Content-Type: application/sdp
Content-Length: 174

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.31 INVITE with Max-Forwards Set to Zero

This is a legal SIP request with the Max-Forwards header set to zero.

A proxy or gateway should not forward the request and respond 483 (Too Many Hops).

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: sip:user@company.com
From: sip:caller@university.edu
Call-ID: 9@10.0.0.1
CSeq: 1 INVITE
Via: SIP/2.0/UDP 135.180.130.133
Max-Forwards: 0
Content-Type: application/sdp
Content-Length: 174

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.32 REGISTER with a Escaped Header in a Legal SIP URL of a Contact

This is a legal REGISTER message where the Contact header contains a SIP URL with an escaped header within it.

Message Details

```
REGISTER sip:company.com SIP/2.0
To: sip:user@company.com
From: sip:user@company.com
Contact: sip:user@host.company.com
Call-ID: k345asrl3fdbv@10.0.0.1
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 135.180.130.133
Contact: <sip:user@example.com?Route=%3Csip:sip.example.com%3E>
```

7.33 REGISTER with a Escaped Header in a Illegal SIP URL of a Contact

This is an illegal message as the REGISTER request contains a SIP

URL with an escaped header but it is not enclosed in <>

A server should respond 400 with an appropriate reason phrase.

Message Details

```
REGISTER sip:company.com SIP/2.0
To: sip:user@company.com
From: sip:user@company.com
Contact: sip:user@host.company.com
Call-ID: k345asrl3fdbv@10.0.0.1
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 135.180.130.133
Contact: sip:user@example.com?Route=%3Csip:sip.example.com%3E
```

7.34 INVITE with Long Values in Headers

This is a legal message that contains long values in many headers.

Message Details

```
INVITE sip:user@company.com SIP/2.0
To: "I have a user name of extreme proportion"
<sip:user@company.com:6000;other-
param=1234567890somethingelselong1234567890>
From: sip:caller@university.edu
Call-ID:
kl24ahsd546folnyt2vbak9sad98u23naodiunzds09a3bqw0sdfbsk34poouymnae004
3nsed09mfkvc74bd0cuwnms05dknw87hjpobd76f
CSeq: 1 INVITE
My-State: sldkjflzdsfaret0803adgaasd0afds0asdaasd
Via: SIP/2.0/UDP sip33.example.com
Via: SIP/2.0/UDP sip32.example.com
Via: SIP/2.0/UDP sip31.example.com
Via: SIP/2.0/UDP sip30.example.com
Via: SIP/2.0/UDP sip29.example.com
Via: SIP/2.0/UDP sip28.example.com
Via: SIP/2.0/UDP sip27.example.com
Via: SIP/2.0/UDP sip26.example.com
Via: SIP/2.0/UDP sip25.example.com
Via: SIP/2.0/UDP sip24.example.com
Via: SIP/2.0/UDP sip23.example.com
Via: SIP/2.0/UDP sip22.example.com
Via: SIP/2.0/UDP sip21.example.com
Via: SIP/2.0/UDP sip20.example.com
Via: SIP/2.0/UDP sip19.example.com
Via: SIP/2.0/UDP sip18.example.com
```

```
Via: SIP/2.0/UDP sip17.example.com
Via: SIP/2.0/UDP sip16.example.com
Via: SIP/2.0/UDP sip15.example.com
Via: SIP/2.0/UDP sip14.example.com
Via: SIP/2.0/UDP sip13.example.com
Via: SIP/2.0/UDP sip12.example.com
Via: SIP/2.0/UDP sip11.example.com
Via: SIP/2.0/UDP sip10.example.com
Via: SIP/2.0/UDP sip9.example.com
Via: SIP/2.0/UDP sip8.example.com
Via: SIP/2.0/UDP sip7.example.com
Via: SIP/2.0/UDP sip6.example.com
Via: SIP/2.0/UDP sip5.example.com
Via: SIP/2.0/UDP sip4.example.com
Via: SIP/2.0/UDP sip3.example.com
Via: SIP/2.0/UDP sip2.example.com
Via: SIP/2.0/UDP sip1.example.com
Via: SIP/2.0/UDP
host.example.com;received=135.180.130.133;branch=C1C3344E2710000000E2
99E568E7potato10potato0potato0
Content-Type: application/sdp

v=0
o=mhandley 29739 7272939 IN IP4 126.5.4.3
s=SIP Call
t=3149328700 0
c=IN IP4 135.180.130.88
m=audio 492170 RTP/AVP 0 12
m=video 3227 RTP/AVP 31
a=rtpmap:31 LPC
```

7.35 OPTIONS with multiple headers.

This is an illegal and badly mangled message.

A server should respond 400 with an appropriate reason phrase if it can. It may just drop this message.

Message Details

```
OPTIONS sip:135.180.130.133 SIP/2.0
Via: SIP/2.0/UDP company.com:5604
From: sip:iuser@company.com
To: sip:user@135.180.130.133
Call-ID: 1804928587@company.com
CSeq: 1 OPTIONS
Expires: 0 0l@company.com
To: sip:user@135.180.130.133
```

```
Call-ID: 1804928587@company.com
CSeq: 1 OPTIONS
Contact: sip:host.company.com
Expires: 0xpires: 0sip:host.company.com
Expires: 0
Contact: sip:host.company.com
```

7.36 INVITE with large number of SDP attributes and telephone subscriber Request-URI

This is a legal message with a large number of SDP attributes and a long telephone subscriber Request-URI

Message Details

```
INVITE sip:+1-972-555-2222;phone-
context=name%40domain;new=user?%22Route%3a%20X%40Y%3bZ=W%22@gw1.wcom.
com;user=phone SIP/2.0
Via: SIP/2.0/UDP iftgw.there.com:5060
From: sip:+1-303-555-1111@ift.here.com;user=phone
To: sip:+1-650-555-2222@ss1.wcom.com;user=phone
Call-ID: 1717@ift.here.com
CSeq: 56 INVITE
Content-Type: application/sdp
Content-Length: 320

v=0
o=faxgw1 2890844527 2890844527 IN IP4 iftgw.there.com
s=Session SDP
c=IN IP4 iftmg.there.com
t=0 0
m=image 49172 udptl t38
a=T38FaxVersion:0
a=T38maxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:260
a=T38FaxUdpEC:t38UDPR
```

7.37 REGISTER with a contact parameter.

This REGISTER contains a contact where the ’user’ parameter should be interpreted
as being a contact-param and not a url-param.

The register should succeed but a subsequent retrieval of the registration must not include "user=phone" as a url-parameter.

Message Details

```
REGISTER sip:bell-tel.com SIP/2.0
Via: SIP/2.0/UDP saturn.bell-tel.com
From: sip:watson@bell-tel.com
To: sip:watson@bell-tel.com
Call-ID: 70710@saturn.bell-tel.com
CSeq: 2 REGISTER
Contact: sip:+1-972-555-2222@gw1.wcom.com;user=phone
```

7.38 REGISTER with a url parameter.

This register contains a contact where the ’user’parameter is a url-param.

The register should succeed and a subsequent retrieval of the registration must include "user=phone" as a url-parameter.

Message Details

```
REGISTER sip:bell-tel.com SIP/2.0
Via: SIP/2.0/UDP saturn.bell-tel.com
From: sip:watson@bell-tel.com
To: sip:watson@bell-tel.com
Call-ID: 70710@saturn.bell-tel.com
CSeq: 3 REGISTER
Contact: <sip:+1-972-555-2222@gw1.wcom.com;user=phone>
```

7.39 INVITE with an Unquoted Display Name Containing Multiple Tokens

This is a legal INVITE where the To and From header contain display names that contain multiple tokens but are unquoted,

Message Details

```
INVITE sip:t.watson@ieee.org SIP/2.0
Via:     SIP/2.0/UDP c.bell-tel.com
From:    A. Bell <sip:a.g.bell@bell-tel.com>
To:      T. Watson <sip:t.watson@ieee.org>
```

```
Call-ID: 31414@c.bell-tel.com
CSeq:    1 INVITE
```

7.40 INVITE with an Unquoted Display Name Containg Non-Token Characters

This is an illegal invite at the display names in the To and From headers
contain non-token characters but are unquoted.

A server may be intelligent enough to cope with this but may also return a
400 response with an appropriate reason phrase.

Message Details

```
INVITE sip:t.watson@ieee.org SIP/2.0
Via:     SIP/2.0/UDP c.bell-tel.com
From:    Bell, Alexander <sip:a.g.bell@bell-tel.com>
To:      Watson, Thomas <sip:t.watson@ieee.org>
Call-ID: 31415@c.bell-tel.com
CSeq:    1 INVITE
```

7.41 INVITE with Unknown (Higher) Protocol Version in Start Line

This is an illegal INVITE as the SIP Protocol version is unknown.

The server should respond to the request with a bad version error.

Message Details

```
INVITE sip:t.watson@ieee.org SIP/7.0
Via:     SIP/2.0/UDP c.bell-tel.com
From:    A. Bell <sip:a.g.bell@bell-tel.com>
To:      T. Watson <sip:t.watson@ieee.org>
Call-ID: 31417@c.bell-tel.com
CSeq:    1 INVITE
```

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