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Session Initiation Protocol PSTN Call Flows

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Abstract

This informational document gives examples of Session Initiation Protocol (SIP) call flows showing interworking with the Public Switched Telephone Network (PSTN). Elements in these call flows include SIP User Agents and Clients, SIP Proxy Servers, and PSTN Gateways. Scenarios include SIP to PSTN, PSTN to SIP, and PSTN to PSTN via SIP. 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). Call flow diagrams and message details are shown.

Conventions used in this document

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC-2119 [1].

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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.

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

These call flows are based on the current version 2.0 of SIP in RFC 3261[2] with SDP usage described in RFC 3264[3].

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

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. For full information about SIP to ISUP mapping, refer to [4].

Basic SIP call flow examples contained in a companion document, RFC yyyy[5].

1.1 General Assumptions

A number of architecture, network, and protocol assumptions underly 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 aid in the understanding of the call flow examples.

The authentication of SIP User Agents in these example call flows is performed using SIP Digest as defined in [3] and [6].

Some 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.

These flows show UDP for transport. Other transport schemes could also be used.

Throughout this document the call flows show a network where the proxy servers authenticate users on behalf of gateways. Gateways may also authenticate users directly. Both of these are reasonable usages of SIP. If gateways do not authenticate directly they would be to refuse requests from entities other than trusted proxy servers with which they have effective channel security (for example [7] or [8])."

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.

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.

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.

Telephone numbers are usually represented as SIP URIs. Note that an alternative is the use of the tel URI [9].

1.2 Legend for Message Flows

Dashed lines (---) represent signaling messages that are mandatory to the call scenario. These messages can be SIP or PSTN signaling. The arrow indicates the direction of message flow.

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

Messages with parentheses around their name represent optional messages.

Messages are identified in the Figures as F1, F2, etc. This references the message details in the list that follows the Figure. Comments in the message details are shown in the following form:

/* Comments. */

1.3 SIP Protocol Assumptions

This document is informational only and is NOT NORMATIVE in any sense.

For simplicity in reading and editing the document, there are a number of differences between some of the examples and actual SIP messages. For example, the SIP Digest responses are not actual MD5 encodings. Call-IDs are often repeated, and CSeq counts often begin at 1. Header fields are usually shown in the same order. Usually

only the minimum required header field set is shown, others that would normally be present such as Accept, Supported, Allow, etc are not shown.

Actors:

Element	Display Name	URI	IP Address
User Agent	BigGuy	UserA@atlanta.com	192.168.100.101
User Agent	LittleGuy	UserB@biloxi.com	192.168.200.201
Proxy Server		ss1.atlanta.com	192.168.255.111
User Agent (Gateway)		gw1.atlanta.com	192.168.255.201
User Agent (Gateway)		gw2.atlanta.com	192.168.255.202
User Agent (Gateway)		gw3.atlanta.com	192.168.255.203
User Agent (Gateway)		ngw1.atlanta.com	192.168.255.101
User Agent (Gateway)		ngw2.atlanta.com	192.168.255.102

2. SIP to PSTN Dialing

In the following scenarios, User A (BigGuy sip:UserA@atlanta.com) is a SIP phone or other SIP-enabled device. User B is reachable via the PSTN at global telephone number +19725552222. 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.

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 and a Gateway.

In the failure scenarios, the call does not complete. In some cases, however, a media stream is still setup. This is due to the fact that some 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.

2.1   Successful SIP to ISUP PSTN call

```
User A           Proxy 1           NGW 1          Switch B
  |                 |                 |                 |
  |   INVITE F1     |                 |                 |
  |---------------->|                 |                 |
  |     100  F2     |                 |                 |
  |<----------------|   INVITE F3     |                 |
  |                 |---------------->|                 |
  |                 |     100  F4     |                 |
  |                 |<----------------|     IAM F5      |
  |                 |                 |---------------->|
  |                 |                 |     ACM F6      |
  |                 |     183 F7      |<----------------|
  |     183 F8      |<----------------|                 |
  |<----------------|                 |                 |
  |        Both 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 +19725552222 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 URI. Note that tel URIs could be used instead of SIP URIs.

User A could use either their SIP address (sip:UserA@atlanta.com) or SIP telephone number (sip:+13145551111@ss1.atlanta.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. For the details of SIP to ISUP mapping, refer to [4].

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Proxy-Authorization: Digest username="UserA", realm="atlanta.com",
 nonce="dc3a5ab25302aa931904ba7d88fa1cf5", opaque="",
 uri="sip:+19725552222@ss1.atlanta.com;user=phone",
 response="ccdca50cb091d587421457305d097458c"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
```

```
Content-Length: 0
```

/* Proxy 1 uses a Location Service function to determine the gateway for terminating this call. 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:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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


F6 ACM User B -> NGW 1

ACM


F7 183 Session Progress NGW 1 -> Proxy 1

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


/* 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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
```

```
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F12 ACK A -> Proxy 1

```
ACK sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F13 ACK Proxy 1 -> NGW 1

```
ACK sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
```

;tag=314159

```
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

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

F14 BYE A -> Proxy 1

```
BYE sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

F15 BYE Proxy 1 -> NGW 1

```
BYE sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
```

```
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

F17 200 OK Proxy 1 -> A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

F18 REL NGW 1 -> B

```
REL
CauseCode=16 Normal
```

F19 RLC B -> NGW 1

```
RLC
```

2.2 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   |
   |                |    180 F8       |<----------------|
   |    180 F9      |<----------------|                 |
   |<---------------|                 |                 |
   |                |                 |  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 a 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 URI.

The host portion of the Request-URI in the INVITE F3 is used to identify the context (customer, trunk group, or line) in which the private number 444-3333 is valid. Otherwise, this INVITE message could get forwarded 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 gateway that serves User C. User C is identified by its extension (444-3333) in the Request-URI sent to GW 1.

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:+19185553333@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sip:UserA@192.168.100.101>
Proxy-Authorization: Digest username="UserA",
 realm="atlanta.com", nonce="qo0dc3a5ab22aa931904badfa1cf5j9h",
 opaque="", uri="sip:+19185553333@ss1.atlanta.com;user=phone",
 response="6c792f5c9fa360358b93c7fb826bf550"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
```

```
CSeq: 2 INVITE
Content-Length: 0
```

F3 INVITE Proxy 1 -> GW 1

```
INVITE sip:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sip:UserA@192.168.100.101>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Content-Length: 0
```

F5 SETUP GW 1 -> User C

```
Protocol discriminator=Q.931
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
Message type=CALL PROC
Channel identification=Exclusive B-channel
```

F7 PROGress User C -> GW 1

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

F8 180 Ringing GW 1 -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sip:4443333@gw1.atlanta.com>
Content-Length: 0
```

F9 180 Ringing Proxy 1 -> User A

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
```

```
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sip:4443333@gw1.atlanta.com>
Content-Length: 0
```

F10 CONNect User C -> GW 1

```
Protocol discriminator=Q.931
Message type=CONN
```

F11 CONNect ACK GW 1 -> User C

```
Protocol discriminator=Q.931
Message type=CONN ACK
```

F12 200 OK GW 1 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sip:4443333@gw1.atlanta.com>
Content-Type: application/sdp
Content-Length: 140

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sip:4443333@gw1.atlanta.com>
Content-Type: application/sdp
Content-Length: 140

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F14 ACK A -> Proxy 1

```
ACK sip:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 ACK
Content-Length: 0
```

F15 ACK Proxy 1 -> GW 1

```
ACK sip:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
```

Call-ID: 2xTb9vxSit55XU7p8@atlanta.com

```
CSeq: 2 ACK
Content-Length: 0


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

F16 BYE A -> Proxy 1

BYE sip:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 3 BYE
Content-Length: 0


F17 BYE Proxy 1 -> GW 1

BYE sip:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
```

```
CSeq: 3 BYE
Content-Length: 0
```

F19 200 OK Proxy 1 -> A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: OtherGuy <sip:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 3 BYE
Content-Length: 0
```

F20 DISConnect GW 1 -> User C

```
Protocol discriminator=Q.931
Message type=DISC
Cause=16 (Normal clearing)
```

F21 RELease User C -> GW 1

```
Protocol discriminator=Q.931
Message type=REL
```

F22 RELease COMplete GW 1 -> User C

```
Protocol discriminator=Q.931
Message type=REL COM
```

2.3 Successful SIP to ISUP PSTN call with overflow

```
User A        Proxy 1         NGW 1          NGW 2        Switch B
  |              |              |              |              |
  |  INVITE F1   |              |              |              |
  |------------->|              |              |              |
  |              |  INVITE F2   |              |              |
  |    100  F3   |------------->|              |              |
  |<-------------|    503 F4    |              |              |
  |              |<-------------|              |              |
  |              |    ACK F5    |              |              |
  |              |------------->|              |              |
  |              |   INVITE F6                 |              |
  |              |---------------------------->|     IAM F7   |
  |              |                             |------------->|
  |              |                             |     ACM F8   |
  |              |            183 F9           |<-------------|
  |   183 F10    |<----------------------------|              |
  |<-------------|                             |              |
  |              Two 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:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Proxy-Authorization: Digest username="UserA",
 realm="atlanta.com", nonce="b59311c3ba05b401cf80b2a2c5ac51b0",
 opaque="", uri="sip:+19725552222@ss1.atlanta.com;user=phone",
 response="ba6ab44923fa2614b28e3e3957789ab0"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Content-Type: application/sdp
Content-Length: 147
```

```
v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 ACK Proxy 1 -> NGW 1

```
ACK sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com>;user=phone>
```

```
 ;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
Content-Length: 0


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

F6 INVITE Proxy 1 -> NGW 2

INVITE sip:+19725552222@ngw2.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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


F8 ACM User B -> NGW 2

ACM


F9 183 Session Progress NGW 2 -> Proxy 1

SIP/2.0 183 Session Progress
```

```
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw2.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw2.atlanta.com
s=-
c=IN IP4 192.168.255.102
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000


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

F10 183 Session Progress Proxy 1 -> User A

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw2.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw2.atlanta.com
s=-
c=IN IP4 192.168.255.102
t=0 0
m=audio 3456 RTP/AVP 0
```

```
a=rtpmap:0 PCMU/8000
```

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.atlanta.com:5060;branch=z9hG4bK2d4790.2
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw2.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw2.atlanta.com
s=-
c=IN IP4 192.168.255.102
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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw2.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141
```

```
v=0
o=GW 2890844527 2890844527 IN IP4 ngw2.atlanta.com
s=-
c=IN IP4 192.168.255.102
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F14 ACK A -> Proxy 1

```
ACK sip:+19725552222@ngw2.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F15 ACK Proxy 1 -> NGW 2

```
ACK sip:+19725552222@ngw2.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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:+19725552222@ngw2.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
```

```
Max-Forwards: 70
Route: <ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

F17 BYE Proxy 1 -> NGW 2

```
BYE sip:+19725552222@ngw2.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.2
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

F19 200 OK Proxy 1 -> User A

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
```

```
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

F20 REL NGW 2 -> B

```
REL
CauseCode=16 Normal
```

F21 RLC B -> NGW 2

```
RLC
```

2.4   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     |<----------------|                |
  |<---------------|                 |                |
  |         Two 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 hangs up, which results in a CANCEL (F9) being sent 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:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Proxy-Authorization: Digest username="UserA",
 realm="atlanta.com", nonce="01cf8311c3b0b2a2c5ac51bb59a05b40",
 opaque="", uri="sip:+19725552222@ss1.atlanta.com;user=phone",
 response="e178fbe430e6680a1690261af8831f40"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
```

```
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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
```

F6 ACM User B -> NGW 1

```
ACM
```

F7 183 Session Progress NGW 1 -> Proxy 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F8 183 Session Progress Proxy 1 -> User A

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
t=0 0
```

```
m=audio 3456 RTP/AVP 0
```

```
a=rtpmap:0 PCMU/8000


/* Caller hears the recorded announcement, then hangs up */

F9 CANCEL A -> Proxy 1

CANCEL sip:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 CANCEL
Content-Length: 0


F10 200 OK Proxy 1 -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 CANCEL
Content-Length: 0


F11 CANCEL Proxy 1 -> NGW 1

CANCEL sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
```

```
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 CANCEL
Content-Length: 0
```

F13 REL NGW 1 -> B

```
REL
CauseCode=18 No user responding
```

F14 RLC B -> NGW 1

```
RLC
```

F15 487 Request Terminated NGW 1 -> Proxy 1

```
SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F16 ACK Proxy 1 -> NGW 1

```
ACK sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F17 487 Request Terminated Proxy 1 -> A

```
SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F18 ACK A -> Proxy 1

```
ACK sip:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

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

```
User A          Proxy 1          NGW 1          Switch B
  |                |                |                |
  |   INVITE F1    |                |                |
  |--------------->|                |                |
  |     100  F2    |                |                |
  |<---------------|   INVITE F3    |                |
  |                |--------------->|                |
  |                |     100  F4    |                |
  |                |<---------------|     IAM F5     |
  |                |                |--------------->|
  |                |                |    REL(1) F6   |
  |                |                |<---------------|
  |                |                |     RLC F7     |
  |                |     404 F8     |--------------->|
  |                |<---------------|                |
  |                |     ACK F9     |                |
  |                |--------------->|                |
  |     404 F10    |                |                |
  |<---------------|                |                |
  |     ACK F11    |                |                |
  |--------------->|                |                |
  |                |                |                |
```

User A calls PSTN User B through a Proxy Server Proxy 1 and a Network Gateway NGW 1. The call is rejected by the PSTN with a ANSI ISUP Release message REL containing a specific Cause code. This cause value (1) is mapped by the Gateway to a SIP 404 Address Incomplete response which is proxied back to User A. For more details of ISUP cause value to SIP responses refer to [4].

Message Details

F1 INVITE A -> Proxy 1

```
INVITE sip:+44-1234@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+44-1234@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Proxy-Authorization: Digest username="UserA",
```

```
   realm="atlanta.com", nonce="j1c3b0b01cf832da2c5ac51bb59a05b40",
```

```
 opaque="", uri="sip:+44-1234@ss1.atlanta.com;user=phone",
 response="a451358d46b55512863efe1dccaa2f42"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+44-1234@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+44-1234@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Content-Type: application/sdp
Content-Length: 147
```

```
v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+44-1234@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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
```

F6 REL User B -> NGW 1

```
REL
CauseValue=1 Unallocated number
```

F7 RLC NGW 1 -> User B

```
RLC
```

/* Network Gateway maps CauseValue=1 to the SIP message 404 Not
   Found */

F8 404 Not Found NGW 1 -> Proxy 1

```
SIP/2.0 404 Not Found
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
```

```
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+44-1234@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Error-Info: <sip:not-found-ann@ann.atlanta.com>
Content-Length: 0
```

F9 ACK Proxy 1 -> NGW 1

```
ACK sip:+44-1234@ngw1.atlanta.com;user=phone SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+44-1234@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F10 404 Not Found Proxy 1 -> User A

```
SIP/2.0 404 Not Found
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+44-1234@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Error-Info: <sip:not-found-ann@ann.atlanta.com>
Content-Length: 0
```

F11 ACK User A -> Proxy 1

```
ACK sip:+44-1234@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+44-1234@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
```

Content-Length: 0

2.6   **Unsuccessful SIP to PSTN: ANM Timeout**

```
User A          Proxy 1           NGW 1           Switch 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:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
```

```
   Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
```

```
CSeq: 1 INVITE
Contact: <sip:UserA@192.168.100.101>
Proxy-Authorization: Digest username="UserA",
 realm="atlanta.com", nonce="da2c5ac51bb59a05j1c3b0b01cf832b40",
 opaque="", uri="sip:+19725552222@ss1.atlanta.com;user=phone",
 response="579cb9db184cdc25bf816f37cbc03c7d"
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F3 INVITE Proxy 1 -> NGW 1

```
INVITE sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
```

```
Contact: <sip:UserA@192.168.100.101>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 client.atlanta.com
s=-
c=IN IP4 192.168.100.101
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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
```

F6 ACM User B -> NGW 1

```
ACM
```

F7 183 Session Progress NGW 1 -> Proxy 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
```

```
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F8 183 Session Progress Proxy 1 -> User A

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19725552222@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* 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=18 No user responding


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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Error-Info: <sip:temp-unavail-ann@ann.atlanta.com>
Content-Length: 0


F12 ACK Proxy 1 -> NGW 1

ACK sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.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 client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.168.100.101
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
```

```
CSeq: 1 INVITE
Error-Info: <sip:temp-unavail-ann@ann.atlanta.com>
Content-Length: 0
```

F14 ACK User A -> Proxy 1

```
ACK sip:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
From: BigGuy <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: LittleGuy <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

3. PSTN to SIP Dialing

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.

To reduce the number of messages, only a single proxy server is shown in these flows, which means that the atlanta.com proxy server has access to the biloxi.com location service.

3.1   Successful PSTN to SIP call

```
   Switch 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
```

```
F2 INVITE A -> Proxy 1

INVITE sip:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserB@192.168.200.201>
Content-Length: 0
```

F6 180 Ringing Proxy 1 -> NGW 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserB@192.168.200.201>
Content-Length: 0
```

F7 ACM NGW 1 -> User A

ACM

F8 200 OK User B -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
Contact: <sip:UserB@192.168.200.201>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 145

v=0
o=UserB 2890844527 2890844527 IN IP4 client.biloxi.com
s=-
c=IN IP4 192.168.200.201
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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserB@192.168.200.201>
Content-Type: application/sdp
Content-Length: 145

v=0
o=UserB 2890844527 2890844527 IN IP4 client.biloxi.com
s=-
c=IN IP4 192.168.200.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F10 ACK NGW 1 -> Proxy 1

```
ACK sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F11 ACK Proxy 1 -> User B

```
ACK sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Max-Forwards: 69
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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

F14 RLC NGW 1 -> User A

RLC

F15 BYE NGW 1-> Proxy 1

```
BYE sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

F16 BYE Proxy 1 -> User B

```
BYE sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Max-Forwards: 69
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
```

```
CSeq: 2 BYE
Content-Length: 0
```

3.2   Successful PSTN to SIP call, Fast Answer

```
Switch 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
```

F2 INVITE NGW 1 -> Proxy 1

```
INVITE sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
```

```
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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@biloxi.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.201
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserB@192.168.200.201>
Content-Type: application/sdp
Content-Length: 145

v=0
o=UserB 2890844527 2890844527 IN IP4 client.biloxi.com
s=-
c=IN IP4 192.168.200.201
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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserB@192.168.200.201>
Content-Type: application/sdp
Content-Length: 145
```

```
v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.200.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F7 ACK NGW 1 -> Proxy 1

```
ACK UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F8 ACK Proxy 1 -> User B

```
ACK UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=130.131.132.14
Max-Forwards: 69
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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

F11 RLC NGW 1 -> User A

RLC

F12 BYE NGW 1 -> Proxy 1

```
BYE sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

F13 BYE Proxy 1 -> User B

```
BYE sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Max-Forwards: 69
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

3.3 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 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 URI used to identify the caller is shown in these flows as sip:551313@gw1.atlanta.com.

Message Details

PBX A -> GW 1

Seizure

GW 1 -> PBX A

Wink

F1 MF Digits PBX A -> GW 1

KP 1 972 555 2222 ST

F2 INVITE GW 1 -> Proxy 1

```
INVITE sip:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
Max-Forwards: 70
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:551313@gw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
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 +19725552222 is located. Based upon location analysis the call is forwarded to SIP User B. */

F3 INVITE Proxy 1 -> User B

```
INVITE sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:551313@gw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.201
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.201
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 1 INVITE
```

```
Contact: <sip:UserB@192.168.200.201>
Content-Length: 0
```

F6 180 Ringing Proxy 1 -> GW 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserB@192.168.200.201>
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
Contact: <sip:UserB@192.168.200.201>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 145

v=0
o=UserB 2890844527 2890844527 IN IP4 client.biloxi.com
s=-
c=IN IP4 192.168.200.201
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.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserB@192.168.200.201>
Content-Type: application/sdp
Content-Length: 145

v=0
o=UserB 2890844527 2890844527 IN IP4 client.biloxi.com
s=-
c=IN IP4 192.168.200.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F9 ACK GW 1 -> Proxy 1

```
ACK sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F10 ACK Proxy 1 -> User B

```
ACK sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Max-Forwards: 69
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 2 BYE
Content-Length: 0


F12 BYE Proxy 1 -> User B

BYE sip:UserB@192.168.200.201 SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Max-Forwards: 69
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
From: <sip:551313@gw1.atlanta.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
```

```
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

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

```
Switch 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
```

F2 INVITE A -> Proxy 1

```
INVITE sip:+1972559999@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@gw1.atlanta.com;user=phone>;tag=076342s
To: <sip:+1972559999@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@gw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 140

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
```

```
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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.201
From: <sip:+13145551111@gw1.atlanta.com;user=phone>;tag=076342s
To: <sip:+1972559999@ss1.atlanta.com;user=phone>;tag=6a34d410
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 1 INVITE
Error-Info: <sip:does-not-exist@ann.atlanta.com>
Content-Length: 0


F4 ACK GW 1 -> Proxy 1

ACK sip:+1972559999@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@gw1.atlanta.com;user=phone>;tag=076342s
To: <sip:+1972559999@ss1.atlanta.com;user=phone>;tag=6a34d410
Call-ID: 4Fde34wkd11wsGFDs3@gw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0


F5 REL GW 1 -> User A

REL
CauseCode=1


F6 RLC User A -> GW 1

RLC
```

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

```
Switch 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 [4].

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
```

F2 INVITE A -> Proxy 1

```
INVITE sip:+19725552222@ngw1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
```

```
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 140

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
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 +19725552222. The call is then forwarded to User B. */

F3 INVITE F3 Proxy 1 -> User B

```
INVITE UserB@biloxi.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.201
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 140

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
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.atlanta.com:5060;branch=z9hG4bKlueha2
```

```
 ;received=192.168.255.201
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.201
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 ACK Proxy 1 -> User B

```
ACK UserB@biloxi.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.201
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F8 ACK NGW 1 -> Proxy 1

```
ACK UserB@biloxi.com SIP/2.0
```

```
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F9 REL NGW 1 -> User A

```
REL
CauseCode=17 Busy
```

F10 RLC User A -> NGW 1

```
RLC
```

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

```
Switch A          NGW 1           Proxy 1          User B
   |                 |                 |                 |
   |     IAM F1      |                 |                 |
   |---------------->|   INVITE F2     |                 |
   |                 |---------------->|   INVITE F3     |
   |                 |     100  F4     |---------------->|
   |                 |<----------------|                 |
   |                 |                 |      600 F5     |
   |                 |                 |<----------------|
   |                 |                 |      ACK F6     |
   |                 |     600 F7      |---------------->|
   |                 |<----------------|                 |
   |                 |     ACK F8      |                 |
   |     ACM F9      |---------------->|                 |
   |<----------------|                 |                 |
   | One 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 and plays busy tone. The caller then disconnects

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
Interworking=encountered
```

F2 INVITE NGW1 -> Proxy 1

```
INVITE sip:+19725552222@ngw1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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 +19725552222. The call is then forwarded to User B. */

F3 INVITE Proxy 1 -> User B

```
INVITE UserB@biloxi.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
```

```
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F6 ACK Proxy 1 -> User B

```
ACK UserB@biloxi.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bKlueha2
```

```
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F8 ACK NGW 1 -> Proxy 1

```
ACK sip:+19725552222@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F9 ACM NGW 1 -> User A

ACM

/* A one 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

F11 RLC NGW 1 -> User A

RLC

3.7 Unsuccessful PSTN->SIP, ACM timeout

```
Switch 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    |                |
  |                |<---------------|                |
```

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.
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
```

F2 INVITE A -> Proxy 1

```
INVITE sip:+19725552222@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
```

```
   To: <sip:+19725552222@ss1.atlanta.com;user=phone>
```

```
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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 +19725552222. The call is then forwarded to User B. */

F3 INVITE Proxy 1 -> User B

```
INVITE sip:UserB@biloxi.com  SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
c c=IN IP4 192.168.255.101
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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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

F11 RLC NGW 1 -> User A

RLC

F12 CANCEL NGW 1 -> Proxy 1

CANCEL sip:+19725552222@ss11.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>

```
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 CANCEL
Content-Length: 0
```

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

```
Switch 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    |                |                |
  |<---------------|                |                |
```

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 message 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
```

F2 INVITE NGW 1 -> Proxy 1

```
INVITE sip:+19725552222@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
```

```
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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 +19725552222. The call is then forwarded to User B. */

F3 INVITE Proxy 1 -> User B

```
INVITE sip:UserB@biloxi.com  SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.201
Max-Forwards: 69
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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
```

F15 RLC NGW 1 -> User A

RLC

3.9   Unsuccessful PSTN->SIP, Caller Abandonment

```
Switch 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
```

F2 INVITE A -> Proxy 1

```
INVITE sip:+19725552222@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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 +19725552222. The call is then forwarded to User B. */

F3 INVITE Proxy 1 -> User B

```
INVITE sip:UserB@biloxi.com  SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141
```

```
v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.201
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:UserB@192.168.200.201>
Content-Length: 0
```

F6 180 Ringing Proxy 1 -> NGW 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
```

```
CSeq: 1 INVITE
Contact: <sip:UserB@192.168.200.201>
Content-Length: 0


F7 ACM NGW 1 -> User A

ACM


/* User A hangs up */

F8 REL User A -> NGW 1

REL
CauseCode=16 Normal


F9 RLC NGW 1 -> User A

RLC


F10 CANCEL NGW 1 -> Proxy 1

CANCEL sip:+19725552222@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 CANCEL
Content-Length: 0


F12 CANCEL Proxy 1 -> User B
```

```
CANCEL sip:UserB@biloxi.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 CANCEL
Content-Length: 0
```

F14 487 Request Terminated User B -> Proxy 1

```
SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F15 ACK Proxy 1 -> User B

```
ACK sip:UserB@biloxi.com SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F16 487 Request Terminated Proxy 1 -> NGW 1

```
SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F17 ACK NGW 1 -> Proxy 1

```
ACK sip:+19725552222@ss11.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

4. PSTN to PSTN 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 [4].

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.

4.1   Successful ISUP PSTN to ISUP PSTN call

```
   Switch A       NGW 1         Proxy 1         GW 2         Switch C
    |              |              |              |              |
    |     IAM F1   |              |              |              |
    |------------->|              |              |              |
    |              |  INVITE F2   |              |              |
    |              |------------->|  INVITE F3   |              |
    |              |              |------------->|    IAM F4    |
    |              |              |              |------------->|
    |              |              |              |    ACM F5    |
    |              |              |   183 F6     |<-------------|
    |              |    183 F7    |<-------------|              |
    |   ACM F8     |<-------------|              |              |
    |<-------------|              |              |              |
    | One Way Voice|     Two 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 Gateway GW 2. The CdPN and CgPN are mapped by GW1 into SIP URIs 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 which is served by GW 2. The Proxy 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 Switch A -> NGW 1

```
IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=918-555-3333,NPI=E.164,NOA=National
```

F2 INVITE NGW 1 -> Proxy 1

```
INVITE sip:+19185553333@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844526 2890844526 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
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:4443333@gw2.atlanta.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKwqwee65
```

```
 ;received=192.168.255.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+13145551111@ngw1.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844526 2890844526 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 192.168.255.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000


F4 IAM GW 2 -> Switch C

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=444-3333,NPI=Private,NOA=Subscriber


F5 ACM Switch C -> GW 2

ACM


/* Based on the ACM 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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:4443333@gw2.atlanta.com>
Content-Type: application/sdp
```

```
Content-Length: 149

v=0
o=GW 987654321 987654321 IN IP4 gw2.atlanta.com
s=-
c=IN IP4 192.168.255.202
t=0 0
m=audio 14918 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F7 183 Session Progress Proxy 1 -> GW 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:4443333@gw2.atlanta.com>
Content-Type: application/sdp
Content-Length: 149

v=0
o=GW 987654321 987654321 IN IP4 gw2.atlanta.com
s=-
c=IN IP4 192.168.255.202
t=0 0
m=audio 14918 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* 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 -> Switch A

ACM

/* User B answers */

F9 ANM Switch C -> GW 2

ANM

F10 200 OK GW 2 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:4443333@gw2.atlanta.com>
Content-Type: application/sdp
Content-Length: 149

v=0
o=GW 987654321 987654321 IN IP4 gw2.atlanta.com
s=-
c=IN IP4 192.168.255.202
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.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:4443333@gw2.atlanta.com>
Content-Type: application/sdp
Content-Length: 149

v=0
o=GW 987654321 987654321 IN IP4 gw2.atlanta.com
s=-
c=IN IP4 192.168.255.202
t=0 0
m=audio 14918 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F12 ANM NGW 1 -> Switch A

```
ANM
```

F13 ACK NGW 1 -> Proxy 1

```
ACK sip:4443333@gw2.atlanta.com SIP/2.0
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F14 ACK Proxy 1 -> GW 2

```
ACK sip:4443333@gw2.atlanta.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.168.255.101
Max-Forwards: 69
From: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.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 Switch C -> GW 2

```
REL
CauseCode=16 Normal
```

F16 BYE GW 2 -> Proxy 1

```
BYE sip:+13145551111@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP gw2.atlanta.com:5060;branch=z9hG4bKtexx6
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
```

```
From: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
To: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 4 BYE
Content-Length: 0
```

F17 RLC GW 2 -> Switch C

RLC

F18 BYE Proxy 1 -> NGW 1

```
BYE sip:+13145551111@ngw1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP gw2.atlanta.com:5060;branch=z9hG4bKtexx6
 ;received=192.168.255.202
Max-Forwards: 69
From: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
To: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP gw2.atlanta.com:5060;branch=z9hG4bKtexx6
 ;received=192.168.255.202
From: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
To: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.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.atlanta.com:5060;branch=z9hG4bKtexx6
 ;received=192.168.255.202
From: <sip:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
To: <sip:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 4 BYE
```

```
Content-Length: 0


F21 REL Switch C -> GW 2

REL
CauseCode=16 Normal


F22 RLC GW 2 -> Switch C

RLC
```

4.2 Successful FGB PBX to ISDN PBX call with overflow

```
 PBX A       GW 1        Proxy 1        GW 2         GW 3        PBX C
   |            |            |            |            |            |
   |  Seizure   |            |            |            |            |
   |----------->|            |            |            |            |
   |    Wink    |            |            |            |            |
   |<-----------|            |            |            |            |
   |MF Digits F1|            |            |            |            |
   |----------->|            |            |            |            |
   |            | INVITE F2  |            |            |            |
   |            |----------->| INVITE F3  |            |            |
   |            |            |----------->|            |            |
   |            |            |   503 F4   |            |            |
   |            |            |<-----------|            |            |
   |            |            |   ACK F5   |            |            |
   |            |            |----------->|            |            |
   |            |            |  INVITE F6              |            |
   |            |            |------------------------>|  SETUP F7  |
   |            |            |          100  F8        |----------->|
   |            |            |<------------------------|CALL PROC F9|
   |            |            |                         |<-----------|
   |            |            |                         | ALERT F10  |
   |            |            |          180 F11        |<-----------|
   |            |  180 F12   |<------------------------|            |
   |            |<-----------|                         |            |
   | Ringtone   |            |                         |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:4443333@ss1.atlanta.com SIP/2.0
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
Max-Forwards: 70
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:551313@gw1.atlanta.com>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844526 2890844526 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
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:4443333@gw2.atlanta.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:551313@gw1.atlanta.com>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844526 2890844526 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
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.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.168.255.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 ACK Proxy 1 -> GW 2

```
ACK sip:4443333@ss1.atlanta.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
```

```
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Max-Forward: 70
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F6 INVITE Proxy 1 -> GW 3

```
INVITE sip:+19185553333@gw3.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:551313@gw1.atlanta.com>
Content-Type: application/sdp
Content-Length: 141

v=0
o=GW 2890844526 2890844526 IN IP4 gw1.atlanta.com
s=-
c=IN IP4 192.168.255.201
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F7 SETUP GW 3 -> PBX C

```
Protocol discriminator=Q.931
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.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F9 CALL PROCeeding PBX C -> GW 3

```
Protocol discriminator=Q.931
Message type=CALL PROC
```

F10 ALERT PBX C -> GW 3

```
Protocol discriminator=Q.931
Message type=PROG
```

/* Based on ALERT message, GW 3 returns a 180 response. */

F11 180 Ringing GW 3 -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
 ;received=192.168.255.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19185553333@gw3.atlanta.com;user=phone>
Content-Length: 0
```

F12 180 Ringing Proxy 1 -> GW 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Record-Route: <sip:ss1.atlanta.com;lr>
```

```
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19185553333@gw3.atlanta.com;user=phone>
Content-Length: 0
```

F13 CONNect PBX C -> GW 3

```
Protocol discriminator=Q.931
Message type=CONN
```

F14 200 OK GW 3 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
 ;received=192.168.255.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 INVITE
Contact: <sip:+19185553333@gw3.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 149

v=0
o=GW 987654321 987654321 IN IP4 gw3.atlanta.com
s=-
c=IN IP4 192.168.255.203
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.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Record-Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 INVITE
```

```
Contact: <sip:+19185553333@gw3.atlanta.com;user=phone>
Content-Type: application/sdp
Content-Length: 149

v=0
o=GW 987654321 987654321 IN IP4 gw3.atlanta.com
s=-
c=IN IP4 192.168.255.203
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:+19185553333@gw3.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F17 ACK Proxy 1 -> GW 3

```
ACK sip:+19185553333@gw3.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.168.255.201
Max-Forwards: 69
From: <sip:551313@gw1.atlanta.com>;tag=63412s
To: <sip:4443333@ss1.atlanta.com>;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F18 CONNect ACK GW 3 -> PBX C

Protocol discriminator=Q.931
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
Message type=DISC
Cause=16 (Normal clearing)
```

F20 BYE GW 3 -> Proxy 1

```
BYE sip:551313@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/UDP gw3.atlanta.com:5060;branch=z9hG4bKkdjuwq
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: <sip:4443333@ss1.atlanta.com>;tag=123456789
To: <sip:551313@gw1.atlanta.com>;tag=63412s
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 BYE
Content-Length: 0
```

F21 BYE Proxy 1 -> GW 1

```
BYE sip:551313@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
Via: SIP/2.0/UDP gw3.atlanta.com:5060;branch=z9hG4bKkdjuwq
 ;received=192.168.255.203
Max-Forwards: 69
From: <sip:4443333@ss1.atlanta.com>;tag=123456789
To: <sip:551313@gw1.atlanta.com>;tag=63412s
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 BYE
Content-Length: 0
```

GW 1 -> PBX A

Seizure removal

F22 RELease GW 3 -> PBX C

```
Protocol discriminator=Q.931
Message type=REL
```

```
F23 200 OK GW 1 -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.2
 ;received=192.168.255.111
Via: SIP/2.0/UDP gw3.atlanta.com:5060;branch=z9hG4bKkdjuwq
 ;received=192.168.255.203
From: <sip:4443333@ss1.atlanta.com>;tag=123456789
To: <sip:551313@gw1.atlanta.com>;tag=63412s
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.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.atlanta.com:5060;branch=z9hG4bKkdjuwq
 ;received=192.168.255.203
From: <sip:4443333@ss1.atlanta.com>;tag=123456789
To: <sip:551313@gw1.atlanta.com>;tag=63412s
Call-ID: 2xTb9vxSit55XU7p8@gw1.atlanta.com
CSeq: 1 BYE
Content-Length: 0


F25 RELease COMplete PBX C -> GW 3

Protocol discriminator=Q.931
Message type=REL COM


PBX A -> GW 1

Seizure removal
```

Security Considerations

Since this document represents NON NORMATIVE examples of SIP session establishment, the security considerations in RFC 3261 [2] apply.

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