SIPPING Working Group
Internet Draft
Document: draft-ietf-sipping-pstn-call-flows-01.txt
Expires: April 2003

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November 2002

Session Initiation Protocol PSTN Call Flows

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Abstract

This document contains best current practice 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, 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), 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 SIP IP communications network. They represent an example minimum set of functionality.

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 implementation of RFC 3261 [2]. These flows represent carefully checked and working group reviewed

scenarios of the most common SIP/PSTN interworking examples as a companion to the specifications.

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]. Other RFCs also comprise the SIP standard but are not used in this set of basic call flows. The SIP/ISUP mapping is based on RFC zzzz [4].

Various PSTN signaling protocols are illustrated in this document: ISDN (Integrated Services Digital Network), ISUP (ISDN User Part) and FGB (Feature Group B) circuit associated signaling. This document shows mainly ANSI ISUP due to its practical origins. However, as used in this document, the usage is virtually identical to the ITU-T International ISUP used as the reference in [4].

Basic SIP call flow examples are contained in a companion document, RFC yyyy [10].

1.1 General Assumptions

A number of architecture, network, and protocol assumptions underlie the call flows in this document. Note that these assumptions are not requirements. They are outlined in this section so that they may be taken into consideration and to 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 [5].

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 TLS, TCP, and UDP for transport. SCTP [6] could also be used. See the discussion in RFC 3261 [2] for details on the transport issues for SIP.

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 [7].

This document shows typical examples of SIP/ISUP interworking. Although in the spirit of the SIP-T framework [8], these examples do not represent a complete implementation of the framework. The examples here represent more of a minimal set of examples for very basic SIP to ISUP interworking, rather than the more complex goal of ISUP transparency. In particular, there are NO examples of encapsulated ISUP in this document. If present, these messages would show S/MIME encryption due to the sensitive nature of this information, as discussed in the SIP-T Framework security considerations section. (Note - RFC 3204 [9] contains an example of an INVITE with encapsulated ISUP.) See the Security Considerations section for a more detailed discussion on the security of these call flows.

In ISUP, the Calling Party Number is abbreviated as CgPN and the Called Party Number is abbreviated as CdPN. Other abbreviations include Numbering Plan Indicator (NPI) and Nature of Address (NOA).

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, in that it does not prescribe the flows that are shown, indeed they MUST NOT be copied due to the reasons described in the next paragraph. On the other hand, these SIP/PSTN call flows represent well-reviewed examples of SIP/PSTN interworking usage that are best common practice according to community consensus.

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	Alice	sip:alice@atlanta.com	192.0.2.101
User Agent	Bob	sip:bob@biloxi.com	192.0.2.200
Proxy Server		sip:ss1.atlanta.com	192.0.2.111
User Agent (Gateway)		sip:gw1.atlanta.com	192.0.2.201
User Agent (Gateway)		sip:gw2.atlanta.com	192.0.2.202
User Agent (Gateway)		sip:gw3.atlanta.com	192.0.2.203
User Agent (Gateway)		sip:ngw1.atlanta.com	192.0.2.103
User Agent (Gateway)		sip:ngw2.atlanta.com	192.0.2.102

Note that NGW 1 and NGW 2 also have a device URIs (Contacts) of sip:ngw1@atlanta.com and sip:ngw2@atlanta.com which resolves to the Proxy Server sip:ss1.wcom.com using DNS SRV records.

2. SIP to PSTN Dialing

In the following scenarios, Alice (Alice sip:alice@atlanta.com) is a SIP phone or other SIP-enabled device. Bob is reachable via the PSTN at global telephone number +19725552222. Alice places a call to Bob through a Proxy Server Proxy 1 and a Network Gateway. In other scenarios, Alice places calls to Carol, 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. Left open is the issue of how the Gateway can determine the accuracy of the telephone number, necessary before passing it as a valid calling party number in the PSTN.

In these scenarios, Alice is a SIP phone or other SIP-enabled device. Alice places a call to Bob in the PSTN or Carol 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 Alice 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

```
Alice           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    |
  |                |                |<---------------|
  |                |                |                |
```

Alice dials the globalized E.164 number +19725552222 to reach Bob. 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.

Alice could use either their SIP address (sip:alice@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 Bob (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, Bob answers the call then Alice disconnects the call. Signaling between NGW 1 and Bob's telephone switch is ANSI ISUP. For the details of SIP to ISUP mapping, refer to [4].

In this flow, notice that the Contact returned by NGW 1 in messages F7-11 is sip:ngw1@atlanta.com. This is because NGW 1 only accepts SIP messages that come through Proxy 1 - any direct signaling will be ignored. Since this Contact URI may be used outside of this dialog and must be routable (Section 8.1.1.8 in RFC 3261 [2]) the Contact URI for NGW 1 must resolve to Proxy 1. This Contact URI is an AOR which resolves via DNS to Proxy 1 (sip:ss1.atlanta.com) which then resolves it to sip:ngw1.atlanta.com which is the address of NGW 1.

This flow shows TCP transport.

Message Details

F1 INVITE Alice -> Proxy 1

```
INVITE sip:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.com;transport=tcp>
Proxy-Authorization: Digest username="alice", realm="atlanta.com",
 nonce="dc3a5ab25302aa931904ba7d88fa1cf5", opaque="",
 uri="sip:+19725552222@ss1.atlanta.com;user=phone",
 response="ccdca50cb091d587421457305d097458c"
Content-Type: application/sdp
Content-Length: 150

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

F2 100 Trying Proxy 1 -> Alice

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 150

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

F4 100 Trying NGW 1 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
```

```
 ;received=192.0.2.111
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 IAM NGW 1 -> Bob

```
IAM
CdPN=972-555-2222,NPI=E.164,NOA=National
CgPN=314-555-1111,NPI=E.164,NOA=National
```

F6 ACM Bob -> NGW 1

```
ACM
```

F7 183 Session Progress NGW 1 -> Proxy 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 ngw1.atlanta.com
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 -> Alice

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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

F9 ANM Bob -> NGW 1

```
ANM
```

F10 200 OK NGW 1 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142
```

```
v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 gw1.atlanta.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F11 200 OK Proxy 1 -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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

F12 ACK Alice -> Proxy 1

```
ACK sip:ngw1@atlanta.com SIP/2.0
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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:ngw1@atlanta.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

/* Alice Hangs Up with Bob. */

F14 BYE Alice -> Proxy 1

```
BYE sip:ngw1@atlanta.com SIP/2.0
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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:ngw1@atlanta.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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

```
Alice            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|
  |                |                |--------------->|
  |                |                |                |
```

Alice is a SIP device while Carol is connected via a Gateway (GW 1) to a PBX. The PBX connection is via a ISDN trunk group. Alice dials Carol'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 Carol. Carolis identified by its extension (444-3333) in the Request-URI sent to GW 1.

Note that the Contact URI for GW1 as used in messages F8, F9, F12, and F13 is sips:4443333@gw1.atlanta.com which does resolve directly to the gateway.

This flow shows the use of Secure SIP (sips) URIs.

Message Details

F1 INVITE Alice -> Proxy 1

```
INVITE sips:+19185553333@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sips:alice@client.atlanta.com>
Proxy-Authorization: Digest username="alice",
 realm="atlanta.com", nonce="qo0dc3a5ab22aa931904badfa1cf5j9h",
 opaque="", uri="sips:+19185553333@ss1.atlanta.com;user=phone",
 response="6c792f5c9fa360358b93c7fb826bf550"
Content-Type: application/sdp
Content-Length: 150

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

F2 100 Trying Proxy 1 -> Alice

```
SIP/2.0 100 Trying
```

```
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Content-Length: 0
```

F3 INVITE Proxy 1 -> GW 1

```
INVITE sips:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sips:ss1.atlanta.com;lr>
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sips:alice@client.atlanta.com>
Content-Type: application/sdp
Content-Length: 150

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

F4 100 Trying GW -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Content-Length: 0
```

F5 SETUP GW 1 -> Carol

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 Carol-> GW 1

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

F7 PROGress Carol-> 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/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sips:ss1.atlanta.com;lr>
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sips:4443333@gw1.atlanta.com>
Content-Length: 0
```

F9 180 Ringing Proxy 1 -> Alice

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sips:ss1.atlanta.com;lr>
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sips:4443333@gw1.atlanta.com>
Content-Length: 0
```

F10 CONNect Carol-> GW 1

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

F11 CONNect ACK GW 1 -> Carol

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

F12 200 OK GW 1 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sips:ss1.atlanta.com;lr>
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sips: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 gw1.atlanta.com
t=0 0
```

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

F13 200 OK Proxy 1 -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sips:ss1.atlanta.com;lr>
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 INVITE
Contact: <sips: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 gw1.atlanta.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F14 ACK Alice -> Proxy 1

```
ACK sips:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sips:ss1.atlanta.com;lr>
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+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 sips:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
```

```
 ;received=192.0.2.101
Max-Forwards: 69
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 2 ACK
Content-Length: 0


/* Alice Hangs Up with Bob. */

F16 BYE Alice -> Proxy 1

BYE sips:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sips:ss1.atlanta.com;lr>
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+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 sips:4443333@gw1.atlanta.com SIP/2.0
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+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/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
```

```
Via: SIP/2.0/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+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/TLS client.atlanta.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sips:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 3 BYE
Content-Length: 0
```

F20 DISConnect GW 1 -> Carol

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

F21 RELease Carol-> GW 1

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

F22 RELease COMplete GW 1 -> Carol

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

2.3 Successful SIP to ISUP PSTN call with overflow

```
Alice          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   |
 |              |                             |<-------------|
 |              |                             |              |
```

Alice calls Bob 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. Bob answers the call. The call is terminated when Alice disconnects the call. NGW 2 and Bob's telephone switch use ANSI ISUP signaling.

NGW 2 also only accepts SIP messages that come through Proxy 1, so

the Contact URI sip:ngw2@atlanta.com is used in this flow.

This flow shows UDP transport.

Message Details

F1 INVITE Alice -> 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: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.com>
Proxy-Authorization: Digest username="alice",
 realm="atlanta.com", nonce="b59311c3ba05b401cf80b2a2c5ac51b0",
 opaque="", uri="sip:+19725552222@ss1.atlanta.com;user=phone",
 response="ba6ab44923fa2614b28e3e3957789ab0"
Content-Type: application/sdp
Content-Length: 150

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

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

F2 INVITE Proxy 1 -> NGW 1

```
INVITE sip:+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=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
```

```
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.com>
Content-Type: application/sdp
Content-Length: 150

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

F3 100 Trying Proxy 1 -> Alice

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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=192.0.2.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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:ngw1@atlanta.com SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.com>
Content-Type: application/sdp
Content-Length: 150

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


F7 IAM NGW 2 -> Bob

IAM
CdPN=972-555-2222,NPI=E.164,NOA=National
CgPN=314-555-1111,NPI=E.164,NOA=National


F8 ACM Bob -> 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=192.0.2.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw2@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

v=0
o=GW 2890844527 2890844527 IN IP4 ngw2.atlanta.com
s=-
c=IN IP4 ngw2.atlanta.com
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 -> Alice

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw2@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

v=0
```

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

F11 ANM Bob -> 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=192.0.2.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw2@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

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

F13 200 OK Proxy 1 -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
```

```
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw2@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

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

F14 ACK Alice -> Proxy 1

```
ACK sip:ngw2@atlanta.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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:ngw2@atlanta.com 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=192.0.2.101
Max-Forwards: 69
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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) */

/* Alice Hangs Up with Bob. */

F16 BYE Alice -> Proxy 1

```
BYE sip:ngw2@atlanta.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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:ngw2@atlanta.com 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=192.0.2.101
Max-Forwards: 69
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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=192.0.2.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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 -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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

```
Alice            Proxy 1           NGW 1            Bob
  |                |                |                |
  |   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    |                |                |
  |--------------->|                |                |
  |                |                |                |
```

Alice calls Bob 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. Alice 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 Alice.)

Message Details

F1 INVITE Alice -> 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: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.com;transport=tcp>
Proxy-Authorization: Digest username="alice",
 realm="atlanta.com", nonce="01cf8311c3b0b2a2c5ac51bb59a05b40",
 opaque="", uri="sip:+19725552222@ss1.atlanta.com;user=phone",
 response="e178fbe430e6680a1690261af8831f40"
Content-Type: application/sdp
Content-Length: 150

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

F2 100 Trying Proxy 1 -> A

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 150

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

F4 100 Trying NGW 1 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 IAM NGW 1 -> Bob

```
IAM
CdPN=972-555-2222,NPI=E.164,NOA=National
CgPN=314-555-1111,NPI=E.164,NOA=National
```

F6 ACM Bob -> 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=192.0.2.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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

F8 183 Session Progress Proxy 1 -> Alice

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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

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

```
a=rtpmap:0 PCMU/8000


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

F9 CANCEL Alice -> 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: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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=192.0.2.111
```

```
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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=192.0.2.111
Via: SIP/2.0/UDP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F18 ACK Alice -> 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: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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

```
Alice            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    |                |                |
  |--------------->|                |                |
  |                |                |                |
```

Alice calls PSTN Bob 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 Alice. For more details of ISUP cause value to SIP response mapping refer to [4].

Message Details

F1 INVITE Alice -> Proxy 1

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

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

```
 opaque="", uri="sip:+44-1234@ss1.atlanta.com;user=phone",
 response="a451358d46b55512863efe1dccaa2f42"
Content-Type: application/sdp
Content-Length: 150

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

F2 100 Trying Proxy 1 -> A

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+44-1234@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 150
```

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

F4 100 Trying NGW 1 -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+44-1234@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 IAM NGW 1 -> Bob

```
IAM
CdPN=44-1234,NPI=E.164,NOA=International
CgPN=314-555-1111,NPI=E.164,NOA=National
```

F6 REL Bob -> NGW 1

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

F7 RLC NGW 1 -> Bob

```
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/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
```

```
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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 -> Alice

```
SIP/2.0 404 Not Found
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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 Alice -> Proxy 1

```
ACK sip:+44-1234@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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

```
Alice           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    |                |                |
  |--------------->|                |                |
```

Alice calls Bob 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 Alice in the SIP network.

Message Details

F1 INVITE Alice -> Proxy 1

```
INVITE sip:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
```

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

```
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.com;transport=tcp>
Proxy-Authorization: Digest username="alice",
 realm="atlanta.com", nonce="da2c5ac51bb59a05j1c3b0b01cf832b40",
 opaque="", uri="sip:+19725552222@ss1.atlanta.com;user=phone",
 response="579cb9db184cdc25bf816f37cbc03c7d"
Content-Type: application/sdp
Content-Length: 150

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

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

F2 100 Trying Proxy 1 -> A

```
SIP/2.0  100 Trying
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
```

```
Contact: <sip:alice@client.atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 150

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

F4 100 Trying NGW 1 -> Proxy 1

```
SIP/2.0  100 Trying
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Content-Length: 0
```

F5 IAM NGW 1 -> Bob

```
IAM
CdPN=972-555-2222,NPI=E.164,NOA=National
CgPN=314-555-1111,NPI=E.164,NOA=National
```

F6 ACM Bob -> NGW 1

```
ACM
```

F7 183 Session Progress NGW 1 -> Proxy 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
```

```
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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

F8 183 Session Progress Proxy 1 -> Alice

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.atlanta.com;lr>
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.atlanta.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.com
CSeq: 1 INVITE
Contact: <sip:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 ngw1.atlanta.com
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 -> Bob

```
REL
```

CauseCode=18 No user responding

F10 RLC Bob -> NGW 1

RLC

F11 480 Temporarily Unavailable NGW 1 -> Proxy 1

```
SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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:ngw1@atlanta.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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 -> Alice

```
SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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 Alice -> Proxy 1

```
ACK sip:+19725552222@ss1.atlanta.com;user=phone SIP/2.0
Max-Forwards: 70
Via: SIP/2.0/TCP client.atlanta.com:5060;branch=z9hG4bK74bf9
From: Alice <sip:+13145551111@ss1.atlanta.com;user=phone>
 ;tag=9fxced76sl
To: Bob <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, Alice is placing calls from the PSTN to Bob in a SIP network. Alice'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           Bob
  |                |                |                |
  |     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, Alice from the PSTN calls Bob through a Network Gateway NGW1 and Proxy Server Proxy 1. When Bob answers the call the media path is setup end-to-end. The call terminates when Alice hangs up the call, with Alice's telephone switch sending an ISUP RELease message which is mapped to a BYE by NGW 1.

Message Details

F1 IAM Alice -> NGW 1

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National

F2 INVITE Alice -> 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:ngw1@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

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

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

F3 INVITE Proxy 1 -> Bob

```
INVITE sip:bob@client.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=192.0.2.103
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:ngw1@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 ngw1.atlanta.com
t=0 0
m=audio 3456 RTP/AVP 0
```

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

F4 100 Trying Bob -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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 Bob -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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:bob@client.biloxi.com>
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=192.0.2.103
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:bob@client.biloxi.com>
Content-Length: 0
```

F7 ACM NGW 1 -> Alice

ACM

F8 200 OK Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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:bob@client.biloxi.com>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 146

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.com
s=-
c=IN IP4 client.biloxi.com
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=192.0.2.103
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:bob@client.biloxi.com>
Content-Type: application/sdp
Content-Length: 146

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

F10 ACK NGW 1 -> Proxy 1

```
ACK sip:bob@client.biloxi.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:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 1 ACK
Content-Length: 0
```

F11 ACK Proxy 1 -> Bob

```
ACK sip:bob@client.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=192.0.2.103
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 Bob -> NGW 1

```
ANM
```

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

/* Alice Hangs Up with Bob. */

F13 REL Alice -> NGW 1

```
REL
CauseCode=16 Normal
```

F14 RLC NGW 1 -> Alice

```
RLC
```

F15 BYE NGW 1-> Proxy 1

```
BYE sip:bob@client.biloxi.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:+19725552222@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.atlanta.com
CSeq: 2 BYE
Content-Length: 0
```

F16 BYE Proxy 1 -> Bob

```
BYE sip:bob@client.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=192.0.2.103
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 Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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=192.0.2.103
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           Bob
  |                |                |                |
  |     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 3.1 except that Bob 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 Alice -> 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/TCP 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:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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

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

F3 INVITE Proxy 1 -> Bob

```
INVITE bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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

F4 100 Trying Proxy 1 -> NGW 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.201
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 Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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:bob@client.biloxi.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 146

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.com
s=-
c=IN IP4 client.biloxi.com
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/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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:bob@client.biloxi.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 147
```

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

F7 ACK NGW 1 -> Proxy 1

```
ACK bob@client.biloxi.com SIP/2.0
Via: SIP/2.0/TCP 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 -> Bob

```
ACK bob@client.biloxi.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP 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 Bob -> NGW 1

```
ANM
```

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

/* Alice Hangs Up with Bob. */

F10 REL ser Alice -> NGW 1

```
REL
CauseCode=16 Normal
```

F11 RLC NGW 1 -> Alice

RLC

F12 BYE NGW 1 -> Proxy 1

```
BYE sip:bob@client.biloxi.com SIP/2.0
Via: SIP/2.0/TCP 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 -> Bob

```
BYE sip:bob@client.biloxi.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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 Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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           Bob
  |                |                |                |
  |    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, Alice dials from PBX A to Bob 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 Bob, GW 1 generates ringing tone for Alice.

Bob answers the call by sending a 200 OK. The call terminates when Alice 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 Alice -> GW 1

Seizure

GW 1 -> PBX A

Wink

F1 MF Digits PBX Alice -> 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: 142

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 gw1.atlanta.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine where the phone number +19725552222 is located. Based upon location analysis the call is forwarded to SIP Bob. */

F3 INVITE Proxy 1 -> Bob

```
INVITE sip:bob@client.biloxi.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=192.0.2.201
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: 142

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

F4 100 Trying Proxy 1 -> GW 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.201
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 Bob -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.201
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:bob@client.biloxi.com>
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=192.0.2.201
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:bob@client.biloxi.com>
Content-Length: 0
```

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

F7 200 OK Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
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:bob@client.biloxi.com>
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 146

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.com
s=-
c=IN IP4 client.biloxi.com
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=192.0.2.201
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:bob@client.biloxi.com>
Content-Type: application/sdp
Content-Length: 146

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

F9 ACK GW 1 -> Proxy 1

```
ACK sip:bob@client.biloxi.com 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 -> Bob

```
ACK sip:bob@client.biloxi.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=192.0.2.201
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) */

/* Alice Hangs Up with Bob. */

F11 BYE GW 1 -> Proxy 1

```
BYE sip:bob@client.biloxi.com 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 -> Bob

```
BYE sip:bob@client.biloxi.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=192.0.2.201
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 Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
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=192.0.2.201
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           Bob
  |                 |                |                |
  |     IAM F1      |                |                |
  |---------------->|   INVITE F2    |                |
  |                 |--------------->|                |
  |                 |     604 F3     |                |
  |                 |<---------------|                |
  |                 |     ACK F4     |                |
  |                 |--------------->|                |
  |     REL F5      |                |                |
  |<----------------|                |                |
  |     RLC F6      |                |                |
  |---------------->|                |                |
  |                 |                |                |
```

Alice 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 Alice -> GW 1

```
IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-9999,NPI=E.164,NOA=National
```

F2 INVITE Alice -> Proxy 1

```
INVITE sip:+1972559999@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/TCP 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;transport=tcp>
Content-Type: application/sdp
Content-Length: 140

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

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

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

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

```
SIP/2.0 604 Does Not Exist Anywhere
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.201
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/TCP 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 -> Alice

```
REL
CauseCode=1
```

F6 RLC Alice -> GW 1

```
RLC
```

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

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

In this scenario, Alice calls Bob through Network Gateway NGW 1 and Proxy 1. The call is routed to Bob by Proxy 1. The call is rejected by Bob 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 Alice's telephone switch. In some scenarios, the busy signal is generated by the Gateway since interworking is indicated. For more discussion on interworking, refer to [4].

Message Details

F1 IAM Alice -> NGW 1

```
IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
```

F2 INVITE Alice -> Proxy 1

```
INVITE sip:+19725552222@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/TCP 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:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 140

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

/* Proxy 1 uses a Location Service function to determine a route for +19725552222. The call is then forwarded to Bob. */

F3 INVITE F3 Proxy 1 -> Bob

```
INVITE bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.201
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:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 140

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

F4 100 Trying Proxy 1 -> NGW 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
```

```
 ;received=192.0.2.201
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 Bob -> Proxy 1

```
SIP/2.0 600 Busy Everywhere
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.201
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 -> Bob

```
ACK bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP 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/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.201
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 bob@biloxi.com SIP/2.0
```

```
Via: SIP/2.0/TCP 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 -> Alice

```
REL
CauseCode=17 Busy
```

F10 RLC Alice -> NGW 1

```
RLC
```

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

```
Switch A          NGW 1           Proxy 1          Bob
  |                |                |                |
  |     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, Alice calls Bob through Network Gateway NGW1 and Proxy 1. The call is routed to Bob by Proxy 1. The call is rejected by the Bob client. NGW 1 sets up a two way voice path to Alice 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 Alice -> 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@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/TCP 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:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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


/* Proxy 1 uses a Location Service function to determine a route for
+19725552222.  The call is then forwarded to Bob. */

F3 INVITE Proxy 1 -> Bob

INVITE bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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

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

F4 100 Trying Bob -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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 Bob -> Proxy 1

```
SIP/2.0 600 Busy Everywhere
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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 -> Bob

```
ACK bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP 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/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
```

```
 ;received=192.0.2.103
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:ngw1@atlanta.com SIP/2.0
Via: SIP/2.0/TCP 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 -> Alice

```
ACM
```

/* A one way speech path is established between NGW 1 and Alice. */

/* Call Released after Alice hangs up. */

F10 REL Alice -> NGW 1

```
REL
CauseCode=16
```

F11 RLC NGW 1 -> Alice

```
RLC
```

3.7 Unsuccessful PSTN->SIP, ACM timeout

```
Switch A          NGW 1           Proxy 1          Bob
  |                |                |                |
  |     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    |                |
  |                |<---------------|                |
```

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

F1 IAM Alice -> NGW 1

IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National

F2 INVITE Alice -> 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:ngw1@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

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

/* Proxy 1 uses a Location Service function to determine a route for +19725552222. The call is then forwarded to Bob. */

F3 INVITE Proxy 1 -> Bob

```
INVITE sip:bob@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=192.0.2.103
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:ngw1@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

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

F4 100 Trying Proxy 1 -> NGW 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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 -> Bob

Same as Message F3

F6 INVITE Proxy 1 -> Bob

Same as Message F3

F7 INVITE Proxy 1 -> Bob

Same as Message F3

F8 INVITE Proxy 1 -> Bob

Same as Message F3

F9 INVITE Proxy 1 -> Bob

Same as Message F3

/* Timer expires in Alice's access network. */

F10 REL Alice -> NGW 1

```
REL
CauseCode=16 Normal
```

F11 RLC NGW 1 -> Alice

```
RLC
```

F12 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
```

F13 200 OK Proxy 1 -> NGW 1

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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     Bob
  |                |                |                |
  |     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, Alice calls Bob 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. Bob does not respond with 180 Ringing. Alice's network disconnects the call with a release REL (CauseCode=102 Timeout).

Message Details

F1 IAM Alice -> 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:ngw1@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

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

/* Proxy 1 uses a Location Service function to determine a route for +19725552222. The call is then forwarded to Bob. */

F3 INVITE Proxy 1 -> Bob

```
INVITE sip:bob@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=192.0.2.201
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:ngw1@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

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

F4 INVITE NGW 1 -> Proxy 1

Same as Message F2

F5 INVITE Proxy 1 -> Bob

Same as Message F3

F6 INVITE NGW 1 -> Proxy 1

Same as Message F2

F7 INVITE Proxy 1 -> Bob

Same as Message F3

F8 INVITE NGW 1 -> Proxy 1

Same as Message F2

F9 INVITE Proxy 1 -> Bob

Same as Message F3

F10 INVITE NGW 1 -> Proxy 1

Same as Message F2

F11 INVITE Proxy 1 -> Bob

Same as Message F3

F12 INVITE NGW 1 -> Proxy 1

Same as Message F2

F13 INVITE Proxy 1 -> Bob

Same as Message F3

/* A timer expires in Alice's access network. */

F14 REL Alice -> NGW 1

REL
CauseCode=102 Timeout

F15 RLC NGW 1 -> Alice

RLC

3.9 Unsuccessful PSTN->SIP, Caller Abandonment

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

F1 IAM Alice -> NGW 1

```
IAM
CgPN=314-555-1111,NPI=E.164,NOA=National
CdPN=972-555-2222,NPI=E.164,NOA=National
```

F2 INVITE Alice -> Proxy 1

```
INVITE sip:+19725552222@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/TCP 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:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142

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

/* Proxy 1 uses a Location Service function to determine a route for +19725552222. The call is then forwarded to Bob. */

F3 INVITE Proxy 1 -> Bob

```
INVITE sip:bob@biloxi.com  SIP/2.0
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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:ngw1@atlanta.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 142
```

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

F4 100 Trying Bob -> Proxy 1

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.201
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 Bob -> Proxy 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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:bob@client.biloxi.com;transport=tcp>
Content-Length: 0
```

F6 180 Ringing Proxy 1 -> NGW 1

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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:bob@client.biloxi.com>
Content-Length: 0
```

F7 ACM NGW 1 -> Alice

ACM

/* Alice hangs up */

F8 REL Alice -> NGW 1

REL
CauseCode=16 Normal

F9 RLC NGW 1 -> Alice

RLC

F10 CANCEL NGW 1 -> Proxy 1

```
CANCEL sip:+19725552222@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/TCP 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/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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 -> Bob

```
CANCEL sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP 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 Bob -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
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 Bob -> Proxy 1

```
SIP/2.0 487 Request Terminated
Via: SIP/2.0/TCP ss1.atlanta.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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 -> Bob

```
ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP 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/TCP ngw1.atlanta.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
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@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/TCP 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, Alice in the PSTN calls Carol who is an extension on a PBX. Alice's telephone switch signals via SS7 to the Network Gateway NGW 1, while Carol'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

Carol 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 Alice can hear the ringing tone generated by PBX C.

Carol answers the call and the media path is cut through in both directions. Bob hangs up terminating the call.

Message Details

F1 IAM Switch Alice -> 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 sips:+19185553333@ss1.atlanta.com;user=phone  SIP/2.0
Via: SIP/2.0/TLS ngw1.atlanta.com:5061;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sips:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sips:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sips:ngw1@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

v=0
o=GW 2890844526 2890844526 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 ngw1.atlanta.com
t=0 0
m=audio 3456 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 sips:4443333@gw2.atlanta.com SIP/2.0
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS ngw1.atlanta.com:5061;branch=z9hG4bKwqwee65
```

```
 ;received=192.0.2.103
Max-Forwards: 69
Record-Route: <sips:ss1.atlanta.com;lr>
From: <sips:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sips:+19185553333@ss1.atlanta.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sips:ngw1@atlanta.com>
Content-Type: application/sdp
Content-Length: 142

v=0
o=GW 2890844526 2890844526 IN IP4 ngw1.atlanta.com
s=-
c=IN IP4 ngw1.atlanta.com
t=0 0
m=audio 3456 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 Alice through NGW 1. */

F6 183 Session Progress GW 2 -> Proxy 1

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TLS ngw1.atlanta.com:5061;branch=z9hG4bKlueha2
 ;received=192.0.2.103
Record-Route: <sips:ss1.atlanta.com;lr>
From: <sips:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sips:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sips: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 gw2.atlanta.com
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/TLS ngw1.atlanta.com:5061;branch=z9hG4bKlueha2
 ;received=192.0.2.103
Record-Route: <sips:ss1.atlanta.com;lr>
From: <sips:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sips:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sips: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 gw2.atlanta.com
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

/* Bob answers */

F9 ANM Switch C -> GW 2

ANM

F10 200 OK GW 2 -> Proxy 1

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TLS ngw1.atlanta.com:5061;branch=z9hG4bKlueha2
 ;received=192.0.2.103
Record-Route: <sips:ss1.atlanta.com;lr>
From: <sips:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sips:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sips: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 gw2.atlanta.com
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/TLS ngw1.atlanta.com:5061;branch=z9hG4bKlueha2
 ;received=192.0.2.103
Record-Route: <sips:ss1.atlanta.com;lr>
From: <sips:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sips:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.atlanta.com
CSeq: 1 INVITE
Contact: <sips: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 gw2.atlanta.com
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 sips:4443333@gw2.atlanta.com SIP/2.0
Via: SIP/2.0/TLS ngw1.atlanta.com:5061;branch=z9hG4bKlueha2
Max-Forwards: 70
Route: <sips:ss1.atlanta.com;lr>
From: <sips:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sips:+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 sips:4443333@gw2.atlanta.com SIP/2.0
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS ngw1.atlanta.com:5061;branch=z9hG4bKlueha2
 ;received=192.0.2.103
Max-Forwards: 69
From: <sips:+13145551111@ngw1.atlanta.com;user=phone>;tag=7643kals
To: <sips:+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. */

/* Bob Hangs Up with Alice. */

F15 REL Switch C -> GW 2

```
REL
CauseCode=16 Normal
```

F16 BYE GW 2 -> Proxy 1

```
BYE sips:ngw1@atlanta.com SIP/2.0
Via: SIP/2.0/TLS gw2.atlanta.com:5061;branch=z9hG4bKtexx6
Max-Forwards: 70
Route: <sips:ss1.atlanta.com;lr>
```

```
From: <sips:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
To: <sips:+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 sips:ngw1@atlanta.com SIP/2.0
Via: SIP/2.0/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS gw2.atlanta.com:5061;branch=z9hG4bKtexx6
 ;received=192.0.2.202
Max-Forwards: 69
From: <sips:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
To: <sips:+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/TLS ss1.atlanta.com:5061;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TLS gw2.atlanta.com:5061;branch=z9hG4bKtexx6
 ;received=192.0.2.202
From: <sips:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
To: <sips:+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/TLS gw2.atlanta.com:5061;branch=z9hG4bKtexx6
 ;received=192.0.2.202
From: <sips:+19185553333@ss1.atlanta.com;user=phone>;tag=314159
To: <sips:+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 Alice calls PBX Carol via Gateway GW 1 and Proxy 1. During the attempt to reach Carol 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 Carol answering the call. The call is terminated when Carol hangs up.

Message Details

PBX Alice -> GW 1

Seizure

GW 1 -> PBX A

Wink

F1 MF Digits PBX Alice -> 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 gw1.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

/* Proxy 1 uses a Location Service function to determine where B is located. 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=192.0.2.201
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 gw1.atlanta.com
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=192.0.2.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
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=192.0.2.201
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=192.0.2.201
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 gw1.atlanta.com
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=192.0.2.201
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=192.0.2.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
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=192.0.2.201
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=192.0.2.111
Via: SIP/2.0/UDP gw1.atlanta.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
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 gw3.atlanta.com
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=192.0.2.201
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 gw3.atlanta.com
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=192.0.2.201
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. */

/* Bob Hangs Up with Alice. */

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=192.0.2.203
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=192.0.2.111
Via: SIP/2.0/UDP gw3.atlanta.com:5060;branch=z9hG4bKkdjuwq
 ;received=192.0.2.203
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=192.0.2.203
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 Alice -> GW 1

Seizure removal

Security Considerations

This document provides examples of mapping from SIP to ISUP and ISUP to SIP. The gateways in these examples are compliant with the Security Considerations Section of RFC zzzz [4] which is summarized here.

There are few security concerns relating to the mapping of ISUP to SIP besides privacy considerations in the calling party number passing. Some concerns relating to the mapping from tel URI parameters to ISUP including the user creation of parameters and codes relating to called number and local number portability (LNP). An operator of a gateway should use policies similar to those present in PSTN switches to avoid security problems.

The mapping from a SIP response code to an ISUP Cause Code presents a theoretical risk, so a gateway operator may implement policies controlling this mapping. Gateways should also not rely on the contents of the From header field for identity information, as it may be arbitrarily populated by a user. Instead, some sort of cryptographic authentication and authorization should be used for identity determination. These flows show both HTTP Digest for authentication of users, although for brevity the challenge is not always shown.

The early media cut-through shown in some flows is another potential security risk, but it is also required for proper interaction with the PSTN. Again, a gateway operator should use proper policies relating to early media to prevent fraud and misuse. Finally, a user agent (even a properly authenticated one) can launch multiple simultaneous requests through a gateway, constituting a denial of service attack. The adoption of policies to limit the number of simultaneous requests from a single entity may be used to prevent this attack.

As discussed in the SIP-T framework [8] SIP/ISUP interworking can be employed as an interdomain signaling mechanism that may be subject to pre-existing trust relationships between administrative domains. Any administrative domain implementing SIP-T or SIP/ISUP interworking should have an adequate security apparatus (including elements that manage any appropriate policies to manage fraud and billing in an interdomain environment) in place to ensure that the translation of ISUP information does not result in any security violations.

Although no examples of this are shown in this document, transporting ISUP in SIP bodies may provide opportunities for abuse, fraud, and privacy concerns, especially when SIP-T requests can be generated, inspected or modified by arbitrary SIP endpoints. ISUP MIME bodies should be secured (preferably with S/MIME as detailed in RFC 3261 [2]) to alleviate this concern. Authentication properties provided by S/MIME would allow the recipient of a SIP-T message to ensure that the ISUP MIME body was generated by an authorized entity. Encryption would ensure that only carriers possessing a particular decryption key are capable of inspecting encapsulated ISUP MIME bodies in a SIP request.

Normative References

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Acknowledgments

Thanks to Rohan Mahy, Adam Roach, Gonzalo Camarillo, Cullen Jennings, and Tom Taylor for their detailed comments during the final review. Thanks to Dean Willis for his early contributions to the development of this document. Thanks to Jon Peterson for his help on the security section.

The authors wish to thank Kundan Singh for performing parser validation of messages.

The authors wish to thank the following individuals for their participation in a detailed review of this call flows document: Aseem Agarwal, Rafi Assadi, Ben Campbell, Sunitha Kumar, Jon Peterson, Marc Petit-Huguenin, Vidhi Rastogi, and Bodgey Yin Shaohua.

The authors also wish to thank the following individuals for their assistance: Jean-Francois Mule, Hemant Agrawal, Henry Sinnreich, David Devanatham, Joe Pizzimenti, Matt Cannon, John Hearty, the whole MCI WorldCom IPOP Design team, Scott Orton, Greg Osterhout, Pat Sollee, Doug Weisenberg, Danny Mistry, Steve McKinnon, and Denise Ingram, Denise Caballero, Tom Redman, Ilya Slain, Pat Sollee, John Truetken, and others from MCI WorldCom, 3Com, Cisco, Lucent and Nortel.

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Acknowledgement

Funding for the RFC Editor function is currently provided by the Internet Society.