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                   Session Initiation Protocol (SIP)
             Public Switched Telephone Network (PSTN) Call Flows

Status of this Memo

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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.

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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 of a
   minimum set of functionality.

   It is the hope of the authors that this document will be useful for
   SIP implementers, 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.

Johnston, et al.   Best Current Practice   [Page 2]

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 3398 [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 3665 [10].

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 BCP 14, RFC 2119 [1].

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 HTTP 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 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 a 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 [6].

This document shows typical examples of SIP/ISUP interworking.
Although in the spirit of the SIP-T framework [7], 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 [8] 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 does not prescribe the flows precisely as they are
shown, but rather the flows illustrate the principles for best
practice. They are best practices usages (orderings, syntax,
selection of features for the purpose, handling of error) of SIP
methods, headers and parameters. IMPORTANT: The exact flows here
must not be copied as is by an implementer due to specific incorrect
characteristics that were introduced into the document for
convenience and are listed below. To sum up, the SIP/PSTN call flows
represent well-reviewed examples of SIP usage, which are best common
practice according to IETF 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@a.example.com	192.0.2.101
User Agent	Bob	sip:bob@b.example.com	192.0.2.200
Proxy Server		sip:ss1.a.example.com	192.0.2.111
User Agent (Gateway)		sip:gw1.a.example.com	192.0.2.201
User Agent (Gateway)		sip:gw2.a.example.com	192.0.2.202
User Agent (Gateway)		sip:gw3.a.example.com	192.0.2.203
User Agent (Gateway)		sip:ngw1.a.example.com	192.0.2.103
User Agent (Gateway)		sip:ngw2.a.example.com	192.0.2.102

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

2. SIP to PSTN Dialing

   In the following scenarios, Alice (sip:alice@a.example.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 Alice 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 which is 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         |
      |                   |                 |<---------------|
      |                   |                 |                 |
```
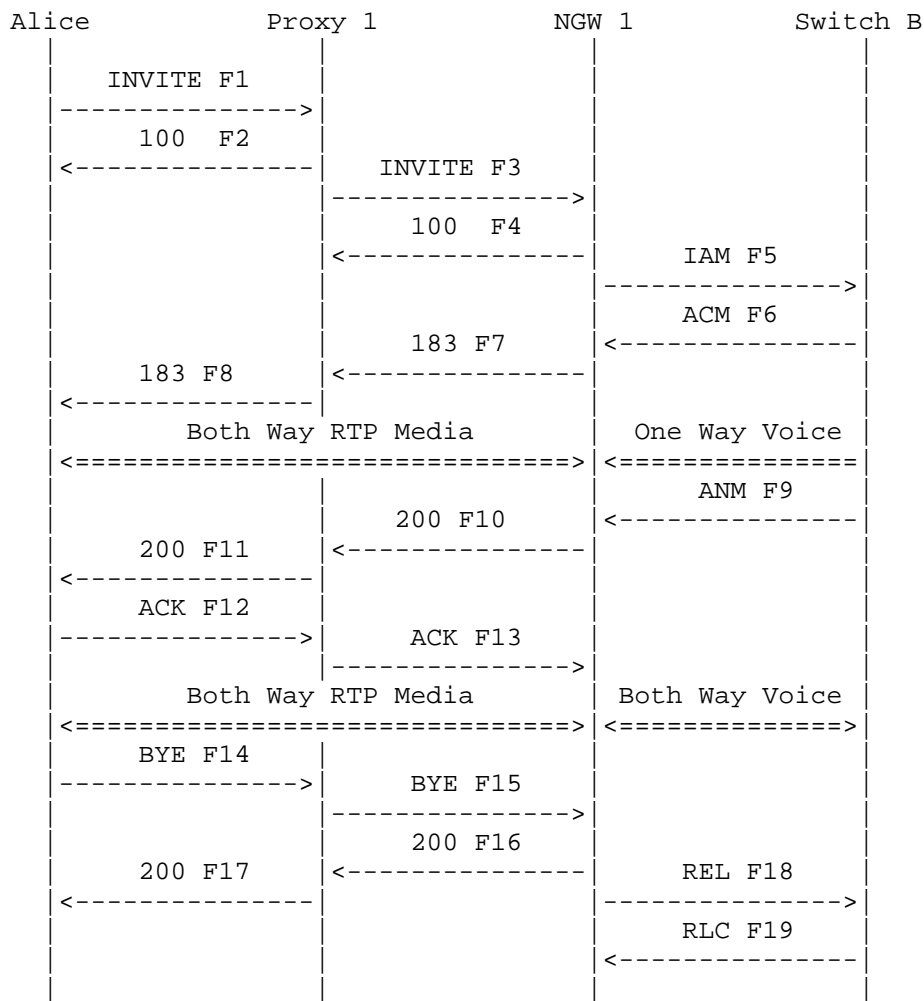
   Alice dials the globalized E.164 number +19725552222 to reach Bob.
   Note that A might have only dialed the last 7 digits, or some other
   dialing plan. It is assumed that the SIP User Agent Client converts
   the digits into a global number and puts them into a SIP URI. Note
   that tel URIs could be used instead of SIP URIs.

   Alice could use either their SIP address (sip:alice@a.example.com) or
   SIP telephone number (sip:+13145551111@ss1.a.example.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@a.example.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 resolves via
DNS to Proxy 1 (sip:ss1.a.example.com) which then resolves it to
sip:ngw1.a.example.com which is the address of NGW 1.

This flow shows TCP transport.

Message Details

F1 INVITE Alice -> Proxy 1

```
INVITE sip:+19725552222@ss1.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.a.example.com;transport=tcp>
Proxy-Authorization: Digest username="alice", realm="a.example.com",
 nonce="dc3a5ab25302aa931904ba7d88fa1cf5", opaque="",
 uri="sip:+19725552222@ss1.a.example.com;user=phone",
 response="ccdca50cb091d587421457305d097458c"
Content-Type: application/sdp
Content-Length: 154

v=0
o=alice 2890844526 2890844526 IN IP4 client.a.example.com
s=-
c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.a.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 154

v=0
o=alice 2890844526 2890844526 IN IP4 client.a.example.com
s=-
c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
```

```
   ;received=192.0.2.111
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:ngw1@a.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 146

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com;transport=tcp>
   Content-Type: application/sdp

```
Content-Length: 146

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 gw1.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:ngw1@a.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 146

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


F12 ACK Alice -> Proxy 1

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

```
   Content-Length: 0


   F13 ACK Proxy 1 -> NGW 1

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

   /* Alice Hangs Up with Bob. */


   F14 BYE Alice -> Proxy 1

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


   F15 BYE Proxy 1 -> NGW 1

   BYE sip:ngw1@a.example.com SIP/2.0
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Max-Forwards: 69
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 BYE
   Content-Length: 0


   F17 200 OK Proxy 1 -> A


   SIP/2.0 200 OK
   Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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. Carol is identified by its extension (444-3333) in the
Request-URI sent to GW 1.

Note that the Contact URI for GW 1, as used in messages F8, F9, F12,
and F13, is sips:4443333@gw1.a.example.com, which resolves 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.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 INVITE
Contact: <sips:alice@client.a.example.com>
Proxy-Authorization: Digest username="alice",
 realm="a.example.com", nonce="qo0dc3a5ab22aa931904badfa1cf5j9h",
 opaque="", uri="sips:+19185553333@ss1.a.example.com;user=phone",
 response="6c792f5c9fa360358b93c7fb826bf550"
Content-Type: application/sdp
Content-Length: 154

v=0
o=alice 2890844526 2890844526 IN IP4 client.a.example.com
s=-
c=IN IP4 client.a.example.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.a.example.com:5061;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Content-Length: 0


   F3 INVITE Proxy 1 -> GW 1

   INVITE sips:4443333@gw1.a.example.com SIP/2.0
   Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Max-Forwards: 69
   Record-Route: <sips:ss1.a.example.com;lr>
   From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Contact: <sips:alice@client.a.example.com>
   Content-Type: application/sdp
   Content-Length: 154

   v=0
   o=alice 2890844526 2890844526 IN IP4 client.a.example.com
   s=-
   c=IN IP4 client.a.example.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.a.example.com:5061;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Content-Length: 0
```
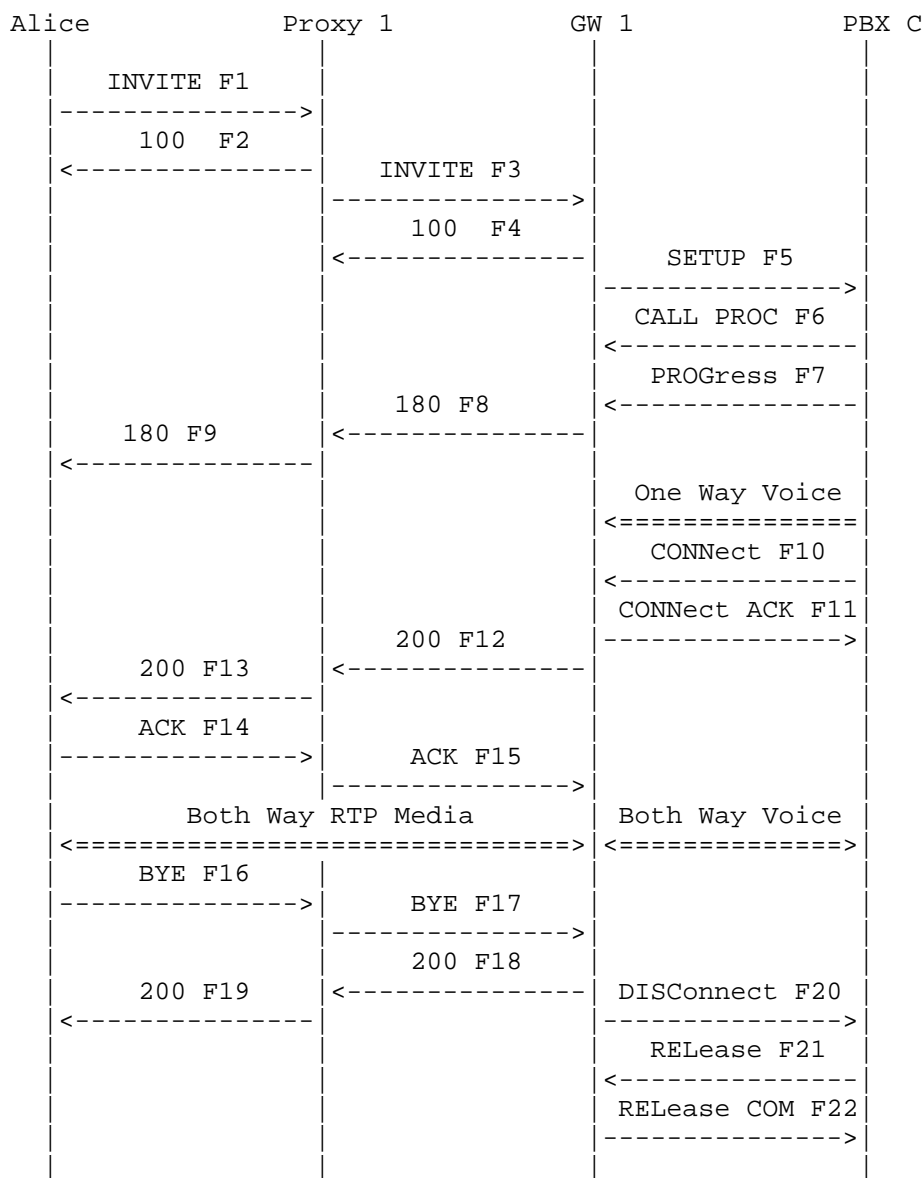
    F5 SETUP GW 1 -> Carol

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

```
F9 180 Ringing Proxy 1 -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sips:ss1.a.example.com;lr>
From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 INVITE
Contact: <sips:4443333@gw1.a.example.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.a.example.com:5061;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sips:ss1.a.example.com;lr>
From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 INVITE
Contact: <sips:4443333@gw1.a.example.com>
Content-Type: application/sdp
Content-Length: 144

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.a.example.com
```

```
   s=-
   c=IN IP4 gw1.a.example.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.a.example.com:5061;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sips:ss1.a.example.com;lr>
   From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Contact: <sips:4443333@gw1.a.example.com>
   Content-Type: application/sdp
   Content-Length: 144

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


   F14 ACK Alice -> Proxy 1

   ACK sips:4443333@gw1.a.example.com SIP/2.0
   Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
   Max-Forwards: 70
   Route: <sips:ss1.a.example.com;lr>
   From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 ACK
   Content-Length: 0
```

      F15 ACK Proxy 1 -> GW 1

      ACK sips:4443333@gw1.a.example.com SIP/2.0
      Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1
      Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
       ;received=192.0.2.101
      Max-Forwards: 69
      From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
       ;tag=9fxced76sl
      To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
       ;tag=314159
      Call-ID: 2xTb9vxSit55XU7p8@a.example.com
      CSeq: 2 ACK
      Content-Length: 0

      /* Alice Hangs Up with Bob. */

      F16 BYE Alice -> Proxy 1

      BYE sips:4443333@gw1.a.example.com SIP/2.0
      Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
      Max-Forwards: 70
      Route: <sips:ss1.a.example.com;lr>
      From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
       ;tag=9fxced76sl
      To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
       ;tag=314159
      Call-ID: 2xTb9vxSit55XU7p8@a.example.com
      CSeq: 3 BYE
      Content-Length: 0

      F17 BYE Proxy 1 -> GW 1

      BYE sips:4443333@gw1.a.example.com SIP/2.0
      Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1
      Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
       ;received=192.0.2.101
      Max-Forwards: 69
      From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
       ;tag=9fxced76sl
      To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
       ;tag=314159
      Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5061;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 3 BYE
Content-Length: 0


F19 200 OK Proxy 1 -> A

SIP/2.0 200 OK
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.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
```
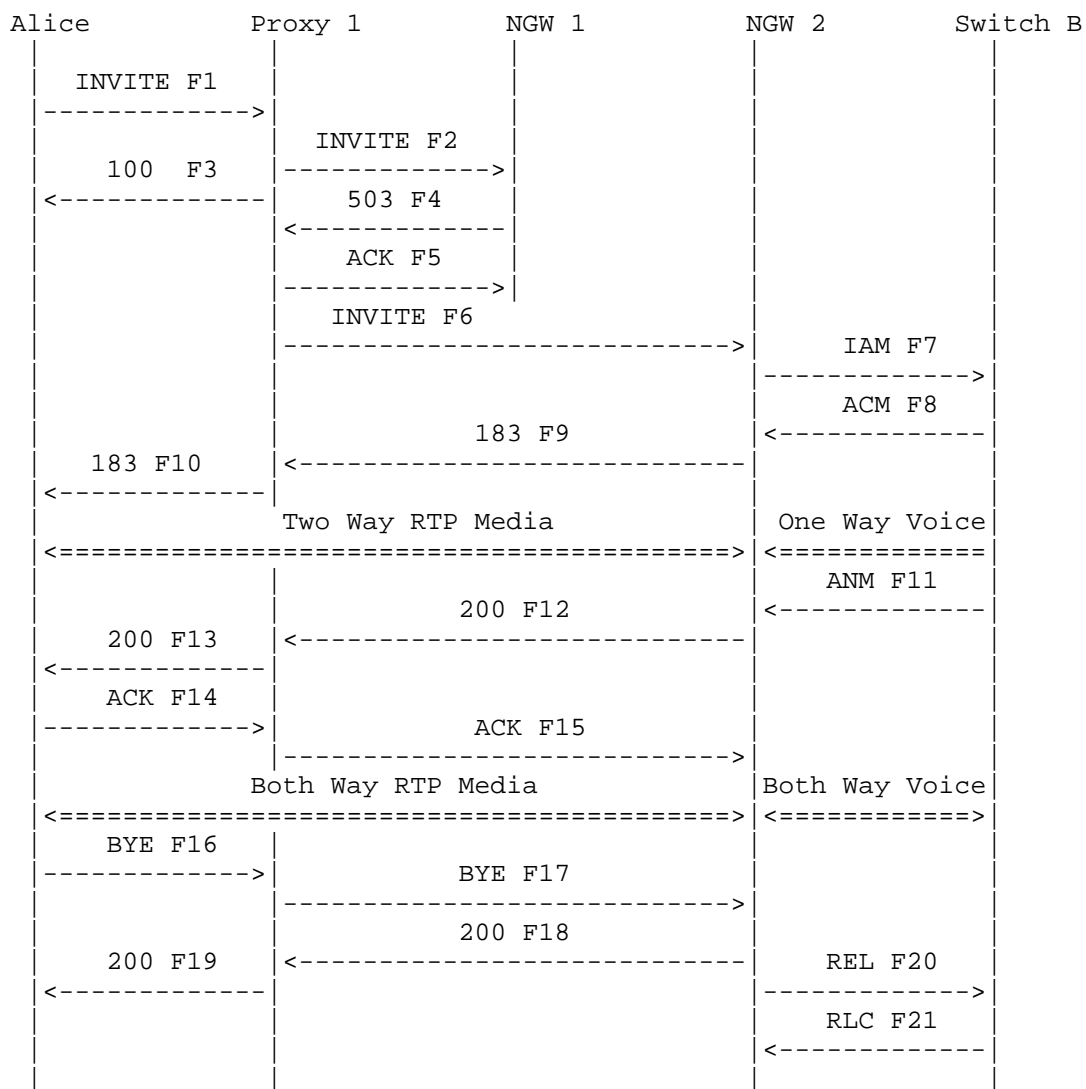
2.3. Successful SIP to ISUP PSTN call with overflow

```
    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@a.example.com is used in this flow.

   This flow shows UDP transport.

   Message Details

   F1 INVITE Alice -> Proxy 1

   INVITE sip:+19725552222@ss1.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
   Max-Forwards: 70
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:alice@client.a.example.com>
   Proxy-Authorization: Digest username="alice",
    realm="a.example.com", nonce="b59311c3ba05b401cf80b2a2c5ac51b0",
    opaque="", uri="sip:+19725552222@ss1.a.example.com;user=phone",
    response="ba6ab44923fa2614b28e3e3957789ab0"
   Content-Type: application/sdp
   Content-Length: 154

   v=0
   o=alice 2890844526 2890844526 IN IP4 client.a.example.com
   s=-
   c=IN IP4 client.a.example.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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Max-Forwards: 69
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl

```
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.a.example.com>
Content-Type: application/sdp
Content-Length: 154

v=0
o=alice 2890844526 2890844526 IN IP4 client.a.example.com
s=-
c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
 ;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Content-Length: 0
```

```
   F5 ACK Proxy 1 -> NGW 1

   ACK sip:ngw1@a.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Max-Forwards: 70
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com>;user=phone>
    ;tag=123456789
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.2
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Max-Forwards: 69
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:alice@client.a.example.com>
   Content-Type: application/sdp
   Content-Length: 154

   v=0
   o=alice 2890844526 2890844526 IN IP4 client.a.example.com
   s=-
   c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.2
    ;received=192.0.2.111
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw2@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw2.a.example.com
   s=-
   c=IN IP4 ngw2.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw2@a.example.com>
   Content-Type: application/sdp

```
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw2.a.example.com
   s=-
   c=IN IP4 ngw2.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.2
    ;received=192.0.2.111
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw2@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw2.a.example.com
   s=-
   c=IN IP4 ngw2.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
```

```
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw2@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

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


   F14 ACK Alice -> Proxy 1

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


   F15 ACK Proxy 1 -> NGW 2

   ACK sip:ngw2@a.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.2
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Max-Forwards: 69
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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@a.example.com SIP/2.0
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
   Max-Forwards: 70
   Route: <ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 BYE
   Content-Length: 0


   F17 BYE Proxy 1 -> NGW 2

   BYE sip:ngw2@a.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.2
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Max-Forwards: 69
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.2
    ;received=192.0.2.111
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
```

```
   ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 BYE
Content-Length: 0


F19 200 OK Proxy 1 -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 BYE
Content-Length: 0


F20 REL NGW 2 -> B

REL
CauseCode=16 Normal


F21 RLC B -> NGW 2

RLC
```

2.4. Successful SIP to SIP using ENUM Query

```
    Alice            DNS Server           Proxy 3              Bob
      |                  |                    |                 |
      |  ENUM Query F1   |                    |                 |
      |--------------->|                      |                 |
      |   Response F2    |                    |                 |
      |<---------------|                      |                 |
      |                 INVITE F3              |                 |
      |----------------------------------->|   INVITE F4        |
      |                  100 F5               |--------------->|
      |<-----------------------------------|     180 F6         |
      |                  180 F7               |<---------------|
      |<-----------------------------------|                    |
      |                                       |     200 F8       |
      |                  200 F9               |<---------------|
      |<-----------------------------------|                    |
      |                  ACK F10              |                 |
      |----------------------------------->|   ACK F11         |
      |                                       |--------------->|
      |                Both Way RTP Media                       |
      |<=====================================================>|
      |                                       |     BYE F12      |
      |                  BYE F13              |<---------------|
      |<-----------------------------------|                    |
      |                  200 F14              |                 |
      |----------------------------------->|     200 F15       |
      |                                       |--------------->|
      |                  |                    |                 |
```
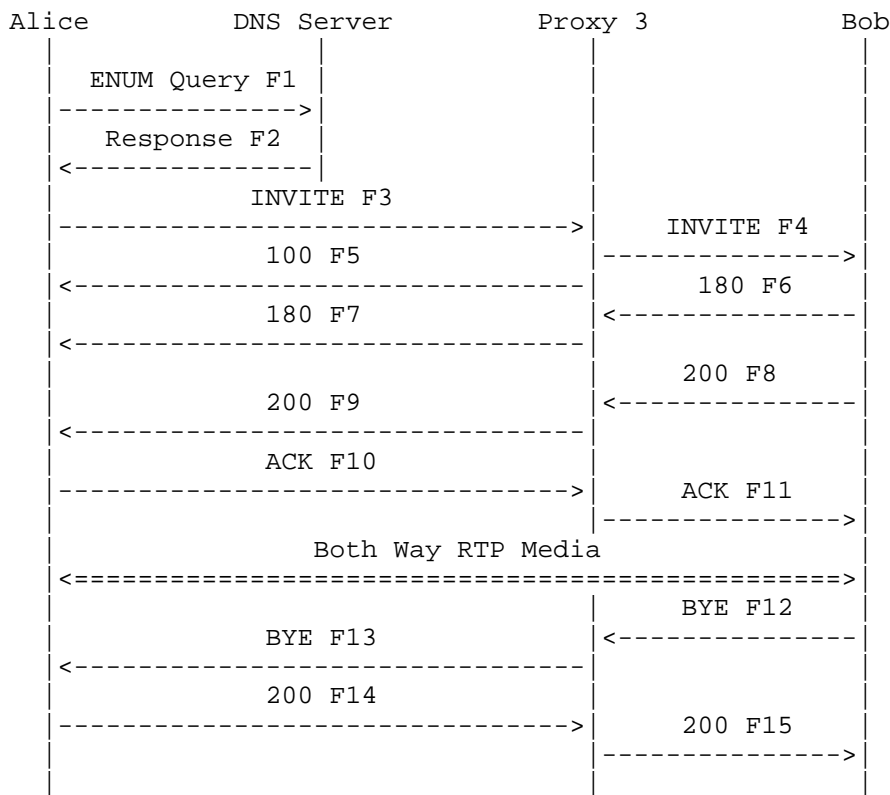
   In this scenario, Alice places a call to Bob by dialing Bob's
   telephone number (9725552222). Alice's UA converts the phone number
   to an E.164 number (+19725552222), and performs an ENUM query [9] on
   the E.164 number (2.2.2.2.5.5.5.2.7.9.1.e164.arpa), which returns a
   NAPTR record containing a SIP AOR URI for Bob
   (sip:+19725552222@b.example.com). As a result, Alice's UA sends an
   INVITE and the call completes over IP bypassing the PSTN.

   The call is terminated when Bob sends a BYE message.

   Message Details

   F1 ENUM Query Alice -> DNS Server

   2.2.2.2.5.5.5.2.7.9.1.e164.arpa

   F2 ENUM NAPTR Set DNS Server -> Alice

   $ORIGIN 2.2.2.2.5.5.5.2.7.9.1.e164.arpa.
         IN NAPTR 100 10 "u" "sip+E2U"
                 "!^.*$!sip:+19725552222@b.example.com!".

   F3 INVITE Alice -> Proxy 3

   INVITE sip:+19725552222@b.example.com SIP/2.0
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
   Max-Forwards: 70
   From: <sip:+13145551111@a.example.com>;tag=9fxced76sl
   To: <tel:+19725552222>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Contact: <sip:+13145551111@client.a.example.com>
   Content-Type: application/sdp
   Content-Length: 154

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

   F4 INVITE Proxy 3 -> Bob

   INVITE sip:+19725552222@client.b.example.com SIP/2.0
   Via: SIP/2.0/UDP ss3.b.example.com:5060;branch=z9hG4bK721e418c4.1
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Max-Forwards: 69
   Record-Route: <sip:ss3.b.example.com;lr>
   From: <sip:+13145551111@a.example.com>;tag=9fxced76sl
   To: <tel:+19725552222>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Contact: <sip:+13145551111@client.a.example.com>
   Content-Type: application/sdp
   Content-Length: 154

   v=0
   o=UserA 2890844526 2890844526 IN IP4 client.a.example.com
   s=-

```
   c=IN IP4 client.a.example.com
   t=0 0
   m=audio 49172 RTP/AVP 0
   a=rtpmap:0 PCMU/8000
```

   F5 100 Trying Proxy 3 -> Alice

```
   SIP/2.0 100 Trying
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   From: <sip:+13145551111@a.example.com>;tag=9fxced76sl
   To: <tel:+19725552222>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Content-Length: 0
```

   F6 180 Ringing B -> Proxy 3

```
   SIP/2.0 180 Ringing
   Via: SIP/2.0/UDP ss3.b.example.com:5060;branch=z9hG4bK721e418c4.1
    ;received=192.0.2.233
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss3.b.example.com;lr>
   From: <sip:+13145551111@a.example.com>;tag=9fxced76sl
   To: <tel:+19725552222>;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Contact: <sip:+19725552222@client.b.example.com>
   Content-Length: 0
```

   F7 180 Ringing Proxy 3 -> Alice

```
   SIP/2.0 180 Ringing
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss3.b.example.com;lr>
   From: <sip:+13145551111@a.example.com>;tag=9fxced76sl
   To: <tel:+19725552222>;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Contact: <sip:+19725552222@client.b.example.com>
   Content-Length: 0
```

   F8 200 OK Bob -> Proxy 3

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss3.b.example.com:5060;branch=z9hG4bK721e418c4.1
    ;received=192.0.2.233
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss3.b.example.com;lr>
   From: <sip:+13145551111@a.example.com>;tag=9fxced76sl
   To: <tel:+19725552222>;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Contact: <sip:+19725552222@client.b.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 151

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

   F9 200 OK Proxy -> Alice

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss3.b.example.com;lr>
   From: <sip:+13145551111@a.example.com>;tag=9fxced76sl
   To: <tel:+19725552222>;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 2 INVITE
   Contact: <sip:+19725552222@client.b.example.com>
   Content-Type: application/sdp
   Content-Length: 151

   v=0
   o=bob 2890844527 2890844527 IN IP4 client.b.example.com
   s=-
   c=IN IP4 192.0.2.100
   t=0 0
   m=audio 3456 RTP/AVP 0
   a=rtpmap:0 PCMU/8000

      F10 ACK Alice -> Proxy 3

      ACK sip:+19725552222@client.b.example.com SIP/2.0
      Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bq9
      Max-Forwards: 70
      Route: <sip:ss3.b.example.com;lr>
      From: <sip:+13145551111@a.example.com>;tag=9fxced76sl
      To: <tel:+19725552222>;tag=314159
      Call-ID: 2xTb9vxSit55XU7p8@a.example.com
      CSeq: 2 ACK
      Content-Length: 0

      F11 ACK Proxy 3 -> Bob

      ACK sip:+19725552222@client.b.example.com SIP/2.0
      Via: SIP/2.0/UDP ss3.b.example.com:5060;branch=z9hG4bK721e418c4.1
      Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bq9
       ;received=192.0.2.101
      Max-Forwards: 69
      From: <sip:+13145551111@a.example.com>;tag=9fxced76sl
      To: <tel:+19725552222>;tag=314159
      Call-ID: 2xTb9vxSit55XU7p8@a.example.com
      CSeq: 2 ACK
      Content-Type: application/sdp
      Content-Length: 0

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

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

      F12 BYE Bob -> Proxy 3

      BYE sip:+13145551111@client.a.example.com SIP/2.0
      Via: SIP/2.0/UDP client.b.example.com:5060;branch=z9hG4bKfgaw2
      Max-Forwards: 70
      Route: <sip:ss3.b.example.com;lr>
      From: <tel:+19725552222>;tag=314159
      To: <sip:+13145551111@a.example.com>;tag=9fxced76sl
      Call-ID: 2xTb9vxSit55XU7p8@a.example.com
      CSeq: 1 BYE
      Content-Length: 0

      F13 BYE Proxy 3 -> Alice

      BYE sip:+13145551111@client.a.example.com SIP/2.0

```
      Via: SIP/2.0/UDP ss3.b.example.com:5060;branch=z9hG4bK721e418c4.1
       ;received=192.0.2.100
      Via: SIP/2.0/UDP client.b.example.com:5060;branch=z9hG4bKfgaw2
      Max-Forwards: 69
      From: <tel:+19725552222>;tag=314159
      To: <sip:+13145551111@a.example.com>;tag=9fxced76sl
      Call-ID: 2xTb9vxSit55XU7p8@a.example.com
      CSeq: 1 BYE
      Content-Length: 0


      F14 200 OK Alice -> Proxy 3


      SIP/2.0 200 OK
      Via: SIP/2.0/UDP ss3.b.example.com:5060;branch=z9hG4bK721e418c4.1
       ;received=192.0.2.233
      Via: SIP/2.0/UDP client.b.example.com:5060;branch=z9hG4bKfgaw2
       ;received=192.0.2.100
      From: <tel:+19725552222>;tag=314159
      To: <sip:+13145551111@a.example.com>;tag=9fxced76sl
      Call-ID: 2xTb9vxSit55XU7p8@a.example.com
      CSeq: 1 BYE
      Content-Length: 0


      F15 200 OK Proxy 3 -> Bob


      SIP/2.0 200 OK
      Via: SIP/2.0/UDP client.b.example.com:5060;branch=z9hG4bKfgaw2
       ;received=192.0.2.100
      From: <tel:+19725552222>;tag=314159
      To: <sip:+13145551111@a.example.com>;tag=9fxced76sl
      Call-ID: 2xTb9vxSit55XU7p8@a.example.com
      CSeq: 1 BYE
      Content-Length: 0
```
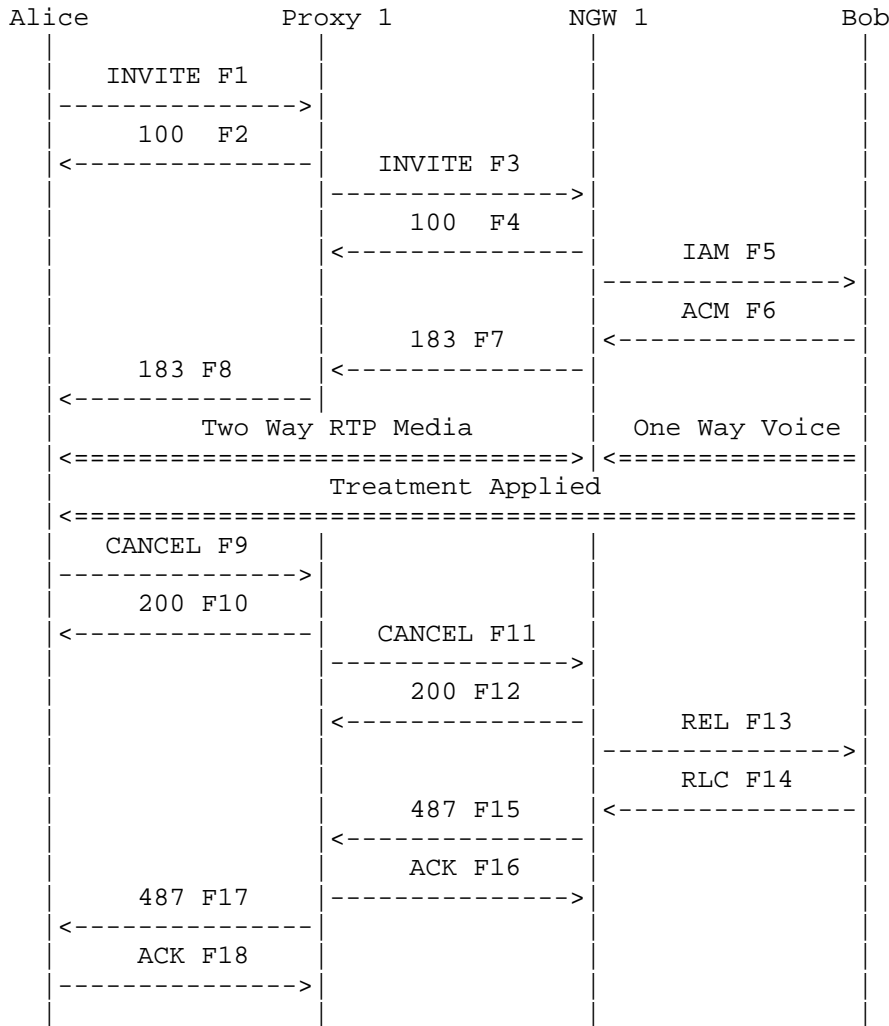
2.5. Unsuccessful SIP to PSTN call: Treatment from PSTN

```
    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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
   Max-Forwards: 70
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:alice@client.a.example.com>
   Proxy-Authorization: Digest username="alice",
    realm="a.example.com", nonce="01cf8311c3b0b2a2c5ac51bb59a05b40",
    opaque="", uri="sip:+19725552222@ss1.a.example.com;user=phone",
    response="e178fbe430e6680a1690261af8831f40"
   Content-Type: application/sdp
   Content-Length: 154

   v=0
   o=alice 2890844526 2890844526 IN IP4 client.a.example.com
   s=-
   c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Max-Forwards: 69
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:alice@client.a.example.com>
   Content-Type: application/sdp
   Content-Length: 154

   v=0
   o=alice 2890844526 2890844526 IN IP4 client.a.example.com
   s=-
   c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

```
v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 ngw1.a.example.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.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 CANCEL
Content-Length: 0


F10 200 OK Proxy 1 -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 CANCEL
Content-Length: 0


F11 CANCEL Proxy 1 -> NGW 1

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

   F16 ACK Proxy 1 -> NGW 1

   ACK sip:+19725552222@ss1.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Max-Forwards: 70
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>

```
   ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Content-Length: 0
```


   F18 ACK Alice -> Proxy 1

```
ACK sip:+19725552222@ss1.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/UDP client.a.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 ACK
Content-Length: 0
```
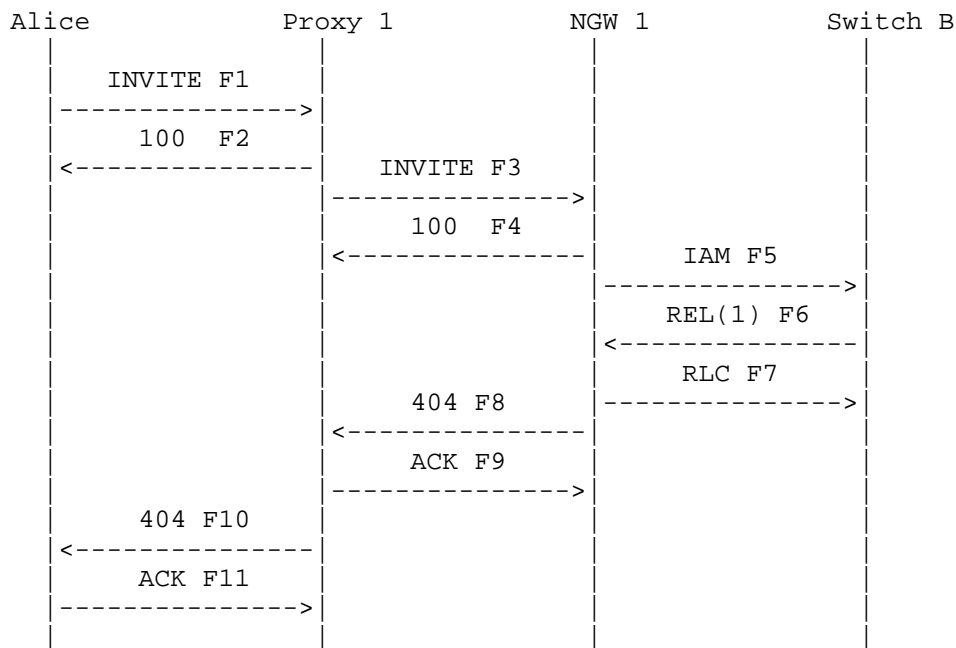
2.6. Unsuccessful SIP to PSTN: REL w/Cause from PSTN

```
     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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
   Max-Forwards: 70
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+44-1234@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:alice@client.a.example.com;transport=tcp>
   Proxy-Authorization: Digest username="alice",
    realm="a.example.com", nonce="j1c3b0b01cf832da2c5ac51bb59a05b40",
    opaque="", uri="sip:+44-1234@ss1.a.example.com;user=phone",

```
   response="a451358d46b55512863efe1dccaa2f42"
Content-Type: application/sdp
Content-Length: 154

v=0
o=alice 2890844526 2890844526 IN IP4 client.a.example.com
s=-
c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+44-1234@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+44-1234@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.a.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 154
```

```
v=0
o=alice 2890844526 2890844526 IN IP4 client.a.example.com
s=-
c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+44-1234@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
     ;received=192.0.2.111
    Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
     ;received=192.0.2.101
    From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
     ;tag=9fxced76sl
    To: Bob <sip:+44-1234@ss1.a.example.com;user=phone>;tag=314159
    Call-ID: 2xTb9vxSit55XU7p8@a.example.com
    CSeq: 1 INVITE
    Error-Info: <sip:not-found-ann@ann.a.example.com>
    Content-Length: 0

    F9 ACK Proxy 1 -> NGW 1

    ACK sip:+44-1234@ngw1.a.example.com;user=phone SIP/2.0
    Max-Forwards: 70
    Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
     ;tag=9fxced76sl
    To: Bob <sip:+44-1234@ss1.a.example.com;user=phone>;tag=314159
    Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
     ;received=192.0.2.101
    From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
     ;tag=9fxced76sl
    To: Bob <sip:+44-1234@ss1.a.example.com;user=phone>;tag=314159
    Call-ID: 2xTb9vxSit55XU7p8@a.example.com
    CSeq: 1 INVITE
    Error-Info: <sip:not-found-ann@ann.a.example.com>
    Content-Length: 0

    F11 ACK Alice -> Proxy 1

    ACK sip:+44-1234@ss1.a.example.com;user=phone SIP/2.0
    Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
    Max-Forwards: 70

```
    From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
     ;tag=9fxced76sl
    To: Bob <sip:+44-1234@ss1.a.example.com;user=phone>;tag=314159
    Call-ID: 2xTb9vxSit55XU7p8@a.example.com
    CSeq: 1 ACK
    Content-Length: 0
```
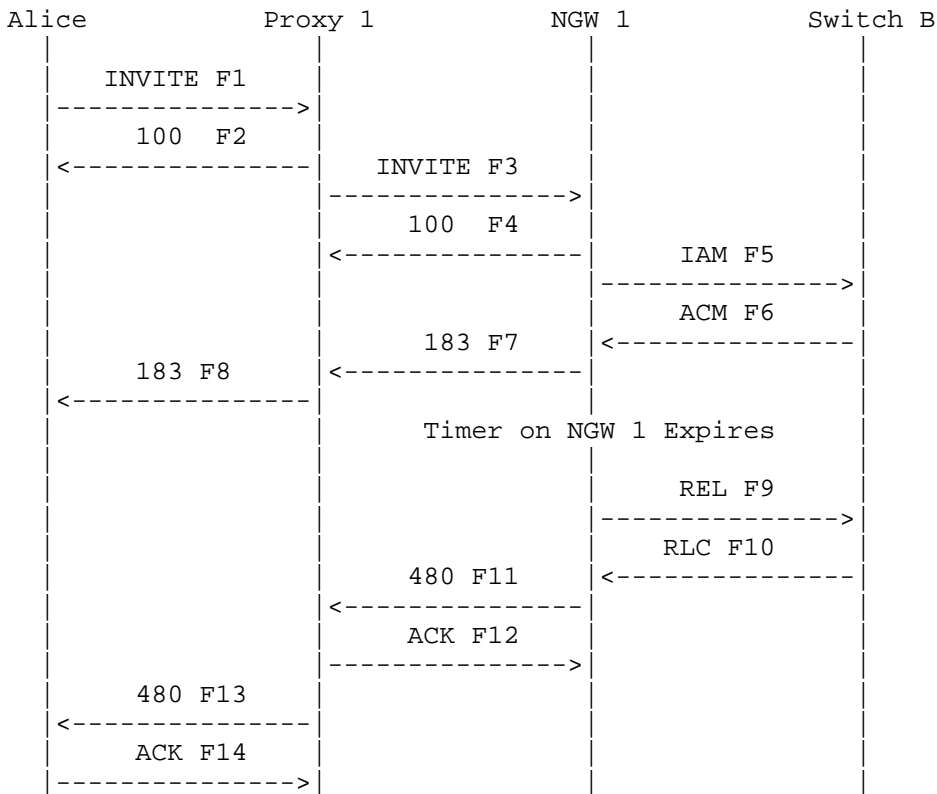
2.7. Unsuccessful SIP to PSTN: ANM Timeout

```
    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.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.a.example.com;transport=tcp>
Proxy-Authorization: Digest username="alice",
 realm="a.example.com", nonce="da2c5ac51bb59a05j1c3b0b01cf832b40",
 opaque="", uri="sip:+19725552222@ss1.a.example.com;user=phone",
 response="579cb9db184cdc25bf816f37cbc03c7d"
Content-Type: application/sdp
Content-Length: 154

v=0
o=alice 2890844526 2890844526 IN IP4 client.a.example.com
s=-
c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Content-Length: 0
```

   F3 INVITE Proxy 1 -> NGW 1

   INVITE sip:+19725552222@ngw1.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Max-Forwards: 69
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:alice@client.a.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 154

   v=0
   o=alice 2890844526 2890844526 IN IP4 client.a.example.com
   s=-
   c=IN IP4 client.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   Record-Route: <sip:ss1.a.example.com;lr>
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 146
```

```
v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
 ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Error-Info: <sip:temp-unavail-ann@ann.a.example.com>
Content-Length: 0
```


F12 ACK Proxy 1 -> NGW 1

```
ACK sip:ngw1@a.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
 ;tag=9fxced76sl
```

```
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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.a.example.com:5060;branch=z9hG4bK74bf9
    ;received=192.0.2.101
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.com
   CSeq: 1 INVITE
   Error-Info: <sip:temp-unavail-ann@ann.a.example.com>
   Content-Length: 0


   F14 ACK Alice -> Proxy 1

   ACK sip:+19725552222@ss1.a.example.com;user=phone SIP/2.0
   Max-Forwards: 70
   Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
   From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>
    ;tag=9fxced76sl
   To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
    ;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@a.example.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 a.example.com proxy server has
access to the b.example.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      |<----------------|
 |                 |<----------------|                  |
 |                 |                  |                  |
```
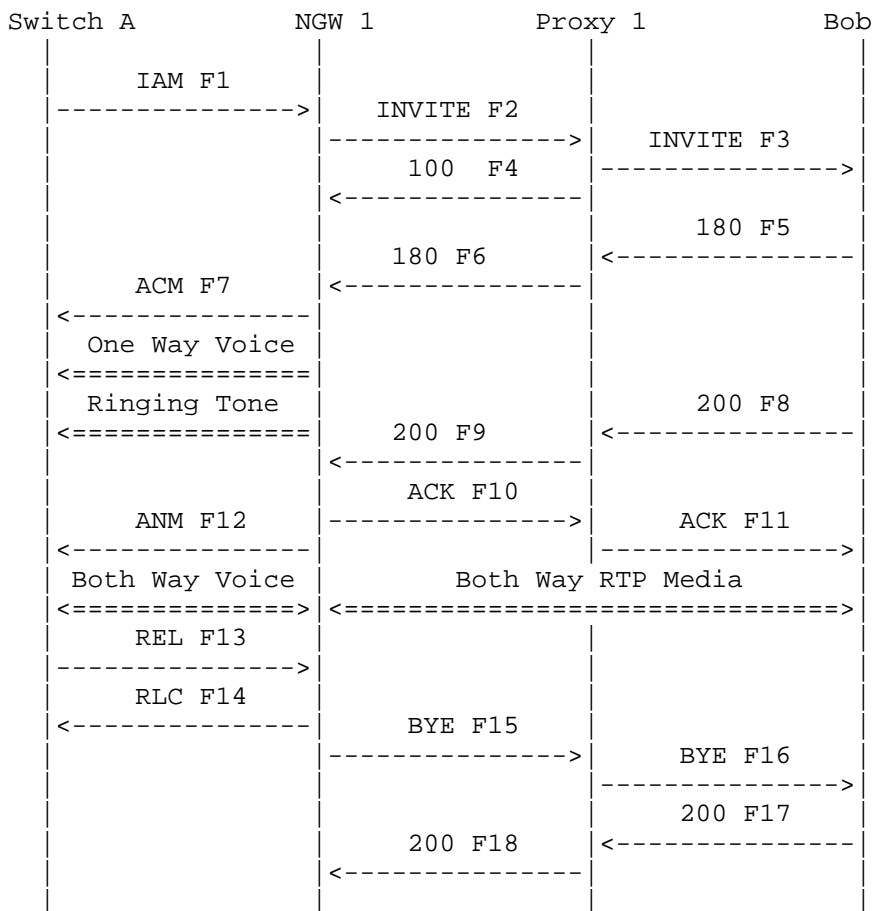
In this scenario, Alice from the PSTN calls Bob through a Network
Gateway NGW1 and Proxy Server Proxy 1. When Bob answers the call,
the media path is setup end-to-end. The call terminates when Alice
hangs up the call, with Alice's telephone switch sending an ISUP
RELease message that is mapped to a BYE 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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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.b.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Max-Forwards: 69
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Content-Length: 0


   F5 180 Ringing Bob -> Proxy 1

   SIP/2.0 180 Ringing
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
```

```
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.com>
   Content-Length: 0
```


```
   F6 180 Ringing Proxy 1 -> NGW 1
```


```
   SIP/2.0 180 Ringing
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   Contact: <sip:bob@client.b.example.com>
   CSeq: 1 INVITE
   Content-Type: application/sdp
   Content-Length: 151
```


```
   v=0
   o=bob 2890844527 2890844527 IN IP4 client.b.example.com
   s=-
   c=IN IP4 client.b.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.com>
   Content-Type: application/sdp
   Content-Length: 151

   v=0
   o=bob 2890844527 2890844527 IN IP4 client.b.example.com
   s=-
   c=IN IP4 client.b.example.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.b.example.com SIP/2.0
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
   Max-Forwards: 70
   Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 ACK
   Content-Length: 0


   F11 ACK Proxy 1 -> Bob

   ACK sip:bob@client.b.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Max-Forwards: 69
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
```

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


   F16 BYE Proxy 1 -> Bob

   BYE sip:bob@client.b.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Max-Forwards: 69
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
```

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


   F17 200 OK Bob -> Proxy 1


   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 2 BYE
   Content-Length: 0
```
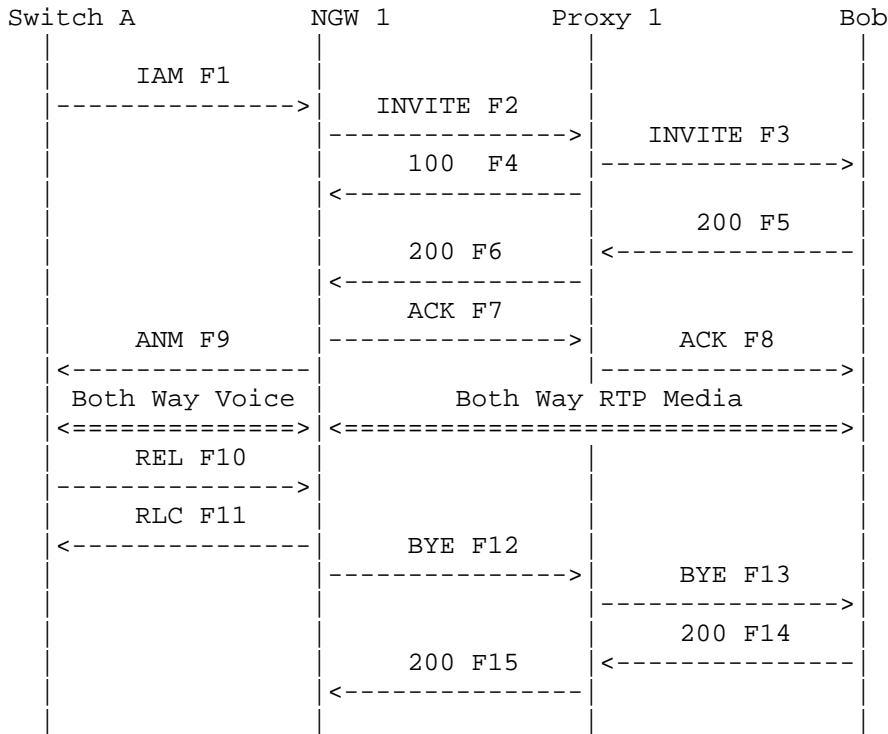
3.2. Successful PSTN to SIP call, Fast Answer

```
   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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2

```
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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@b.example.com SIP/2.0
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Max-Forwards: 69
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.201
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Content-Length: 0

   F5 200 OK Bob -> Proxy 1

   SIP/2.0 200 OK
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 151

   v=0
   o=bob 2890844527 2890844527 IN IP4 client.b.example.com
   s=-
   c=IN IP4 client.b.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.com;transport=tcp>

```
Content-Type: application/sdp
Content-Length: 151

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

F7 ACK NGW 1 -> Proxy 1

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

F8 ACK Proxy 1 -> Bob

```
ACK bob@client.b.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
 ;received=130.131.132.14
Max-Forwards: 69
From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.b.example.com SIP/2.0
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
   Max-Forwards: 70
   Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 2 BYE
   Content-Length: 0

   F13 BYE Proxy 1 -> Bob

   BYE sip:bob@client.b.example.com SIP/2.0
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Max-Forwards: 69
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 2 BYE
   Content-Length: 0

   F14 200 OK Bob -> Proxy 1

   SIP/2.0 200 OK
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 2 BYE
   Content-Length: 0
```
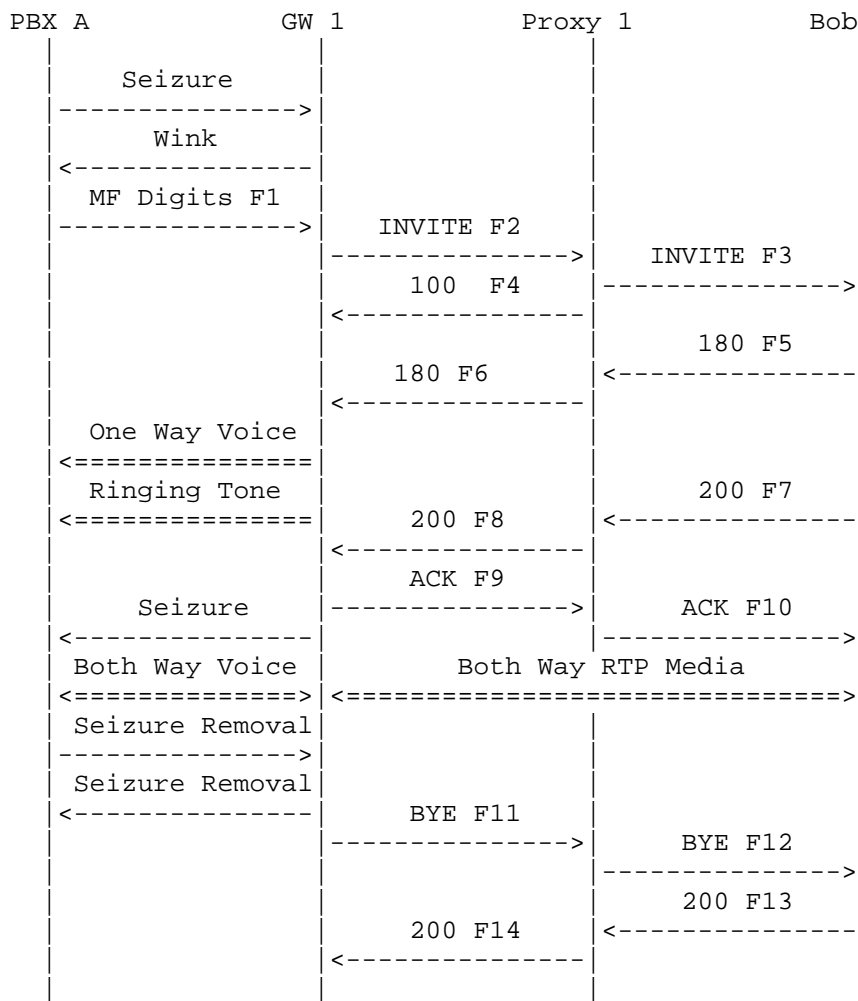
3.3. Successful PBX to SIP call

```
     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 is instead 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 a 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.a.example.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.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
Max-Forwards: 70
From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
To: <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
CSeq: 1 INVITE
Contact: <sip:551313@gw1.a.example.com;user=phone>
Content-Type: application/sdp
Content-Length: 146

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 gw1.a.example.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.b.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
    ;received=192.0.2.201
   Max-Forwards: 69
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:551313@gw1.a.example.com;user=phone>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 gw1.a.example.com
   s=-
   c=IN IP4 gw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.201
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
   CSeq: 1 INVITE
   Content-Length: 0


   F5 180 Ringing Bob -> Proxy 1

   SIP/2.0 180 Ringing
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.201
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
```

```
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.com>
   Content-Length: 0


   F6 180 Ringing Proxy 1 -> GW 1


   SIP/2.0 180 Ringing
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
    ;received=192.0.2.201
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
    ;received=192.0.2.201
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
   Contact: <sip:bob@client.b.example.com>
   CSeq: 1 INVITE
   Content-Type: application/sdp
   Content-Length: 151

   v=0
   o=bob 2890844527 2890844527 IN IP4 client.b.example.com
   s=-
   c=IN IP4 client.b.example.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.a.example.com:5060;branch=z9hG4bKwqwee65
    ;received=192.0.2.201
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.com>
   Content-Type: application/sdp
   Content-Length: 151

   v=0
   o=bob 2890844527 2890844527 IN IP4 client.b.example.com
   s=-
   c=IN IP4 client.b.example.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.b.example.com SIP/2.0
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
   Max-Forwards: 70
   Route: <sip:ss1.a.example.com;lr>
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
   CSeq: 1 ACK
   Content-Length: 0

   F10 ACK Proxy 1 -> Bob

   ACK sip:bob@client.b.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
    ;received=192.0.2.201
   Max-Forwards: 69
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.b.example.com SIP/2.0
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
   Max-Forwards: 70
   Route: <sip:ss1.a.example.com;lr>
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
   CSeq: 2 BYE
   Content-Length: 0


   F12 BYE Proxy 1 -> Bob

   BYE sip:bob@client.b.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
    ;received=192.0.2.201
   Max-Forwards: 69
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
   CSeq: 2 BYE
   Content-Length: 0


   F13 200 OK Bob -> Proxy 1

   SIP/2.0 200 OK
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
    ;received=192.0.2.201
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bKwqwee65
    ;received=192.0.2.201
   From: <sip:551313@gw1.a.example.com;user=phone>;tag=jwdkallkzm
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
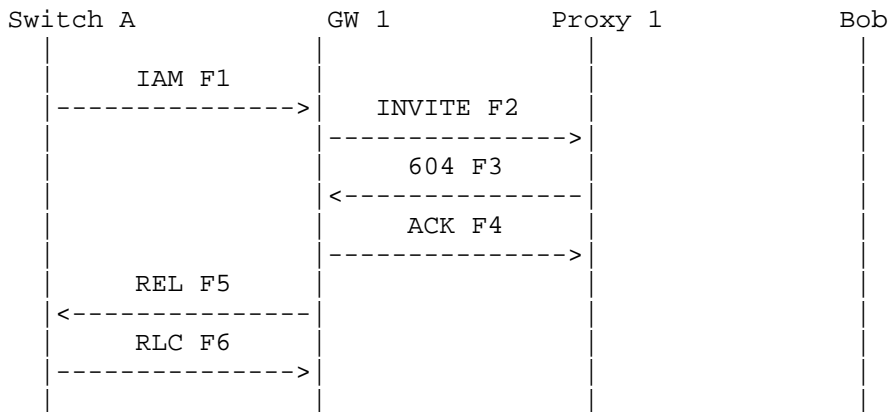   CSeq: 2 BYE
   Content-Length: 0

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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
   Max-Forwards: 70
   From: <sip:+13145551111@gw1.a.example.com;user=phone>;tag=076342s

```
To: <sip:+1972559999@ss1.a.example.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
CSeq: 1 INVITE
Contact:
<sip:+13145551111@gw1.a.example.com;user=phone;transport=tcp>
Content-Type: application/sdp
Content-Length: 144

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.a.example.com
s=-
c=IN IP4 gw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.201
From: <sip:+13145551111@gw1.a.example.com;user=phone>;tag=076342s
To: <sip:+1972559999@ss1.a.example.com;user=phone>;tag=6a34d410
Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
CSeq: 1 INVITE
Error-Info: <sip:does-not-exist@ann.a.example.com>
Content-Length: 0


F4 ACK GW 1 -> Proxy 1

ACK sip:+1972559999@ss1.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@gw1.a.example.com;user=phone>;tag=076342s
To: <sip:+1972559999@ss1.a.example.com;user=phone>;tag=6a34d410
Call-ID: 4Fde34wkd11wsGFDs3@gw1.a.example.com
CSeq: 1 ACK
Content-Length: 0
```
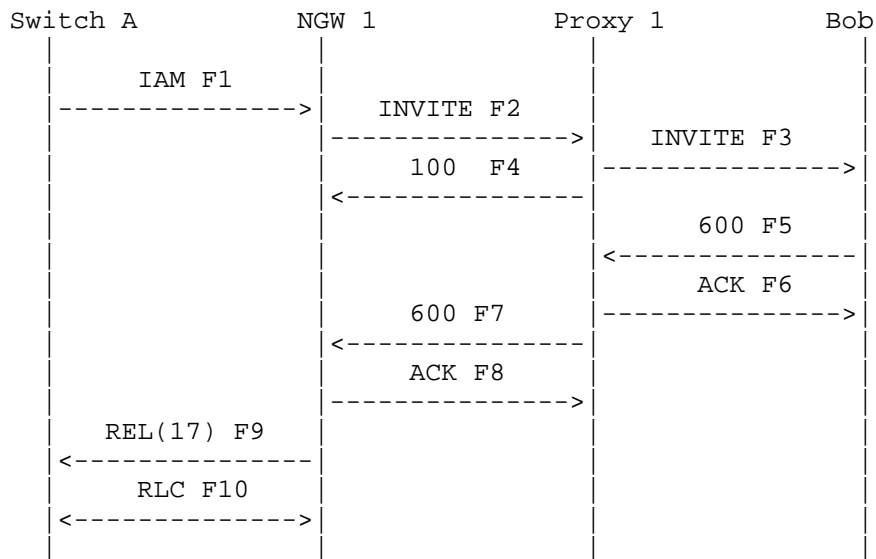
```
    F5 REL GW 1 -> Alice

    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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 144

   v=0
   o=GW 2890844527 2890844527 IN IP4 gw1.a.example.com
   s=-
   c=IN IP4 gw1.a.example.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@b.example.com SIP/2.0
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.201
   Max-Forwards: 69
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com;transport=tcp>
   Content-Type: application/sdp

```
   Content-Length: 144

   v=0
   o=GW 2890844527 2890844527 IN IP4 gw1.a.example.com
   s=-
   c=IN IP4 gw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.201
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.201
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Content-Length: 0
```

   F6 ACK Proxy 1 -> Bob

```
   ACK bob@b.example.com SIP/2.0
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.201
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Content-Length: 0

   F8 ACK NGW 1 -> Proxy 1

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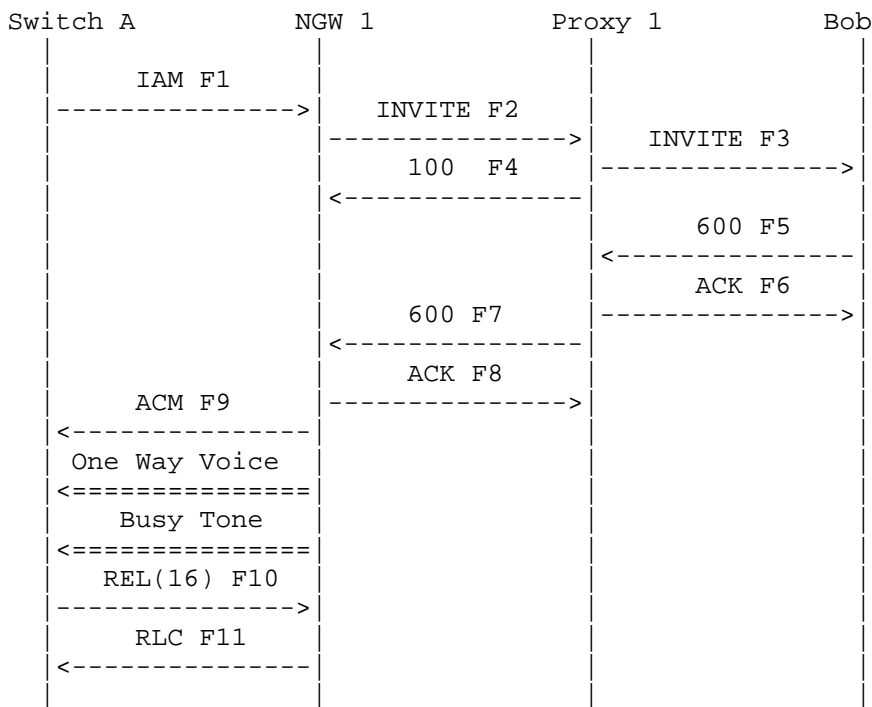   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 ACK
   Content-Length: 0

   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 NGW 1 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.a.example.com;user=phone SIP/2.0
    Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    Max-Forwards: 70
    From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
    To: <sip:+19725552222@ss1.a.example.com;user=phone>
    Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
    CSeq: 1 INVITE
    Contact: <sip:ngw1@a.example.com;transport=tcp>
    Content-Type: application/sdp
    Content-Length: 146

    v=0
    o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
    s=-
    c=IN IP4 ngw1.a.example.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@b.example.com SIP/2.0
    Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
     ;received=192.0.2.103
    Max-Forwards: 69
    Record-Route: <sip:ss1.a.example.com;lr>
    From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
    To: <sip:+19725552222@ss1.a.example.com;user=phone>
    Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
    CSeq: 1 INVITE
    Contact: <sip:ngw1@a.example.com;transport=tcp>
    Content-Type: application/sdp
    Content-Length: 146

    v=0
    o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
    s=-
    c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Content-Length: 0


   F6 ACK Proxy 1 -> Bob

   ACK bob@b.example.com SIP/2.0
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
     ;received=192.0.2.103
    From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
    To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
    Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
    CSeq: 1 INVITE
    Content-Length: 0

    F8 ACK NGW 1 -> Proxy 1

    ACK sip:ngw1@a.example.com SIP/2.0
    Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    Max-Forwards: 70
    From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
    To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
    Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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. */

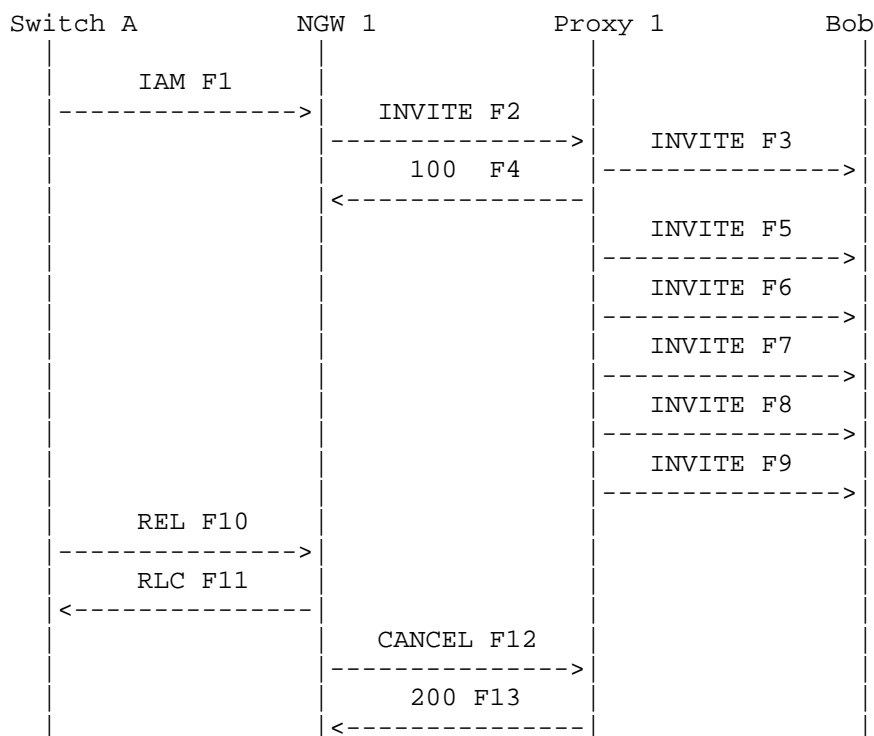
    F10 REL Alice -> NGW 1

    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.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2

```
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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@b.example.com  SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Max-Forwards: 69
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   c c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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
```
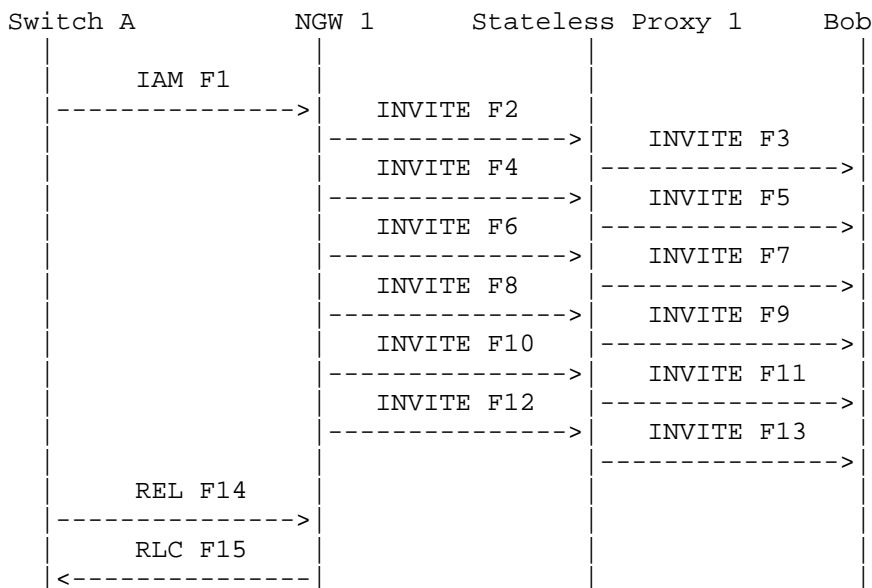
```
F12 CANCEL NGW 1 -> Proxy 1

CANCEL sip:+19725552222@ss1.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE

```
   Contact: <sip:ngw1@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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@b.example.com  SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/UDP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.201
   Max-Forwards: 69
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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    |                             |
   |                |--------------->|                             |
   |                |                |                             |
```
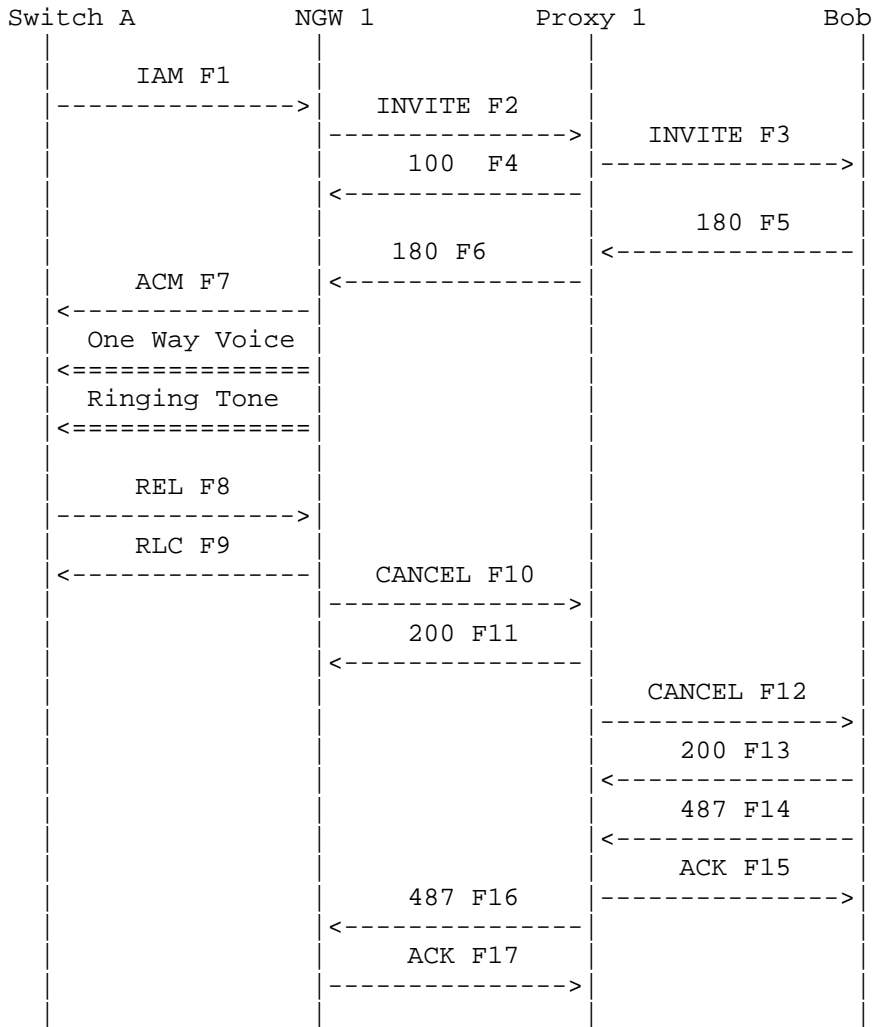
   In this scenario, Alice calls Bob through NGW 1 and Proxy 1. Bob
   does not respond with 200 OK. NGW 1 plays ringing tone since the ACM
   indicates that interworking has been encountered. Alice disconnects
   the call with a Release message REL which is mapped by NGW 1 to a

CANCEL. Note that if Bob had sent a 200 OK response after the REL,
NGW 1 would have sent an ACK and then a BYE to properly terminate the
call.

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.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
CSeq: 1 INVITE
Contact: <sip:ngw1@a.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 146

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 ngw1.a.example.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@b.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
 ;received=192.0.2.103
Max-Forwards: 69
Record-Route: <sip:ss1.a.example.com;lr>
From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
To: <sip:+19725552222@ss1.a.example.com;user=phone>

```
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:ngw1@a.example.com;transport=tcp>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.201
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Content-Length: 0


   F5 180 Ringing Bob -> Proxy 1

   SIP/2.0 180 Ringing
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.com;transport=tcp>
   Content-Length: 0


   F6 180 Ringing Proxy 1 -> NGW 1

   SIP/2.0 180 Ringing
```

```
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sip:ss1.a.example.com;lr>
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:bob@client.b.example.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.a.example.com;user=phone  SIP/2.0
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
```

```
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 CANCEL
   Content-Length: 0


   F12 CANCEL Proxy 1 -> Bob

   CANCEL sip:bob@b.example.com SIP/2.0
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 CANCEL
   Content-Length: 0


   F13 200 OK Bob -> Proxy 1

   SIP/2.0 200 OK
   Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Content-Length: 0


   F15 ACK Proxy 1 -> Bob

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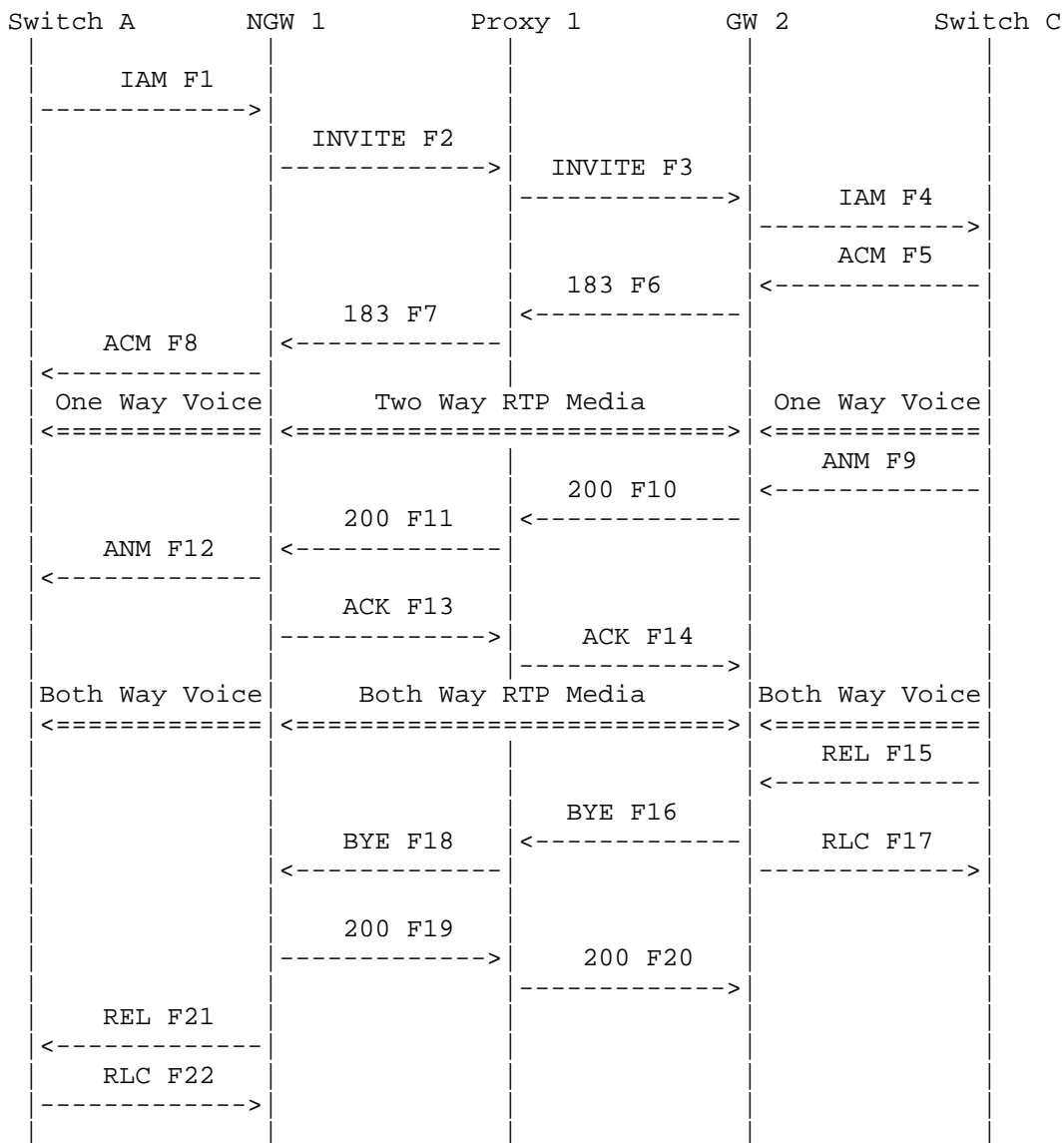
```
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 ACK
   Content-Length: 0
```

```
   F16 487 Request Terminated Proxy 1 -> NGW 1
```

```
   SIP/2.0 487 Request Terminated
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.com
   CSeq: 1 INVITE
   Content-Length: 0
```

```
   F17 ACK NGW 1 -> Proxy 1
```

```
   ACK sip:+19725552222@ss1.a.example.com;user=phone  SIP/2.0
   Via: SIP/2.0/TCP ngw1.a.example.com:5060;branch=z9hG4bKlueha2
   Max-Forwards: 70
   From: <sip:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 4Fde34wkd11wsGFDs3@ngw1.a.example.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 that 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 GW 1 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.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TLS ngw1.a.example.com:5061;branch=z9hG4bKlueha2
Max-Forwards: 70
From: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
To: <sips:+19185553333@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.com
CSeq: 1 INVITE
Contact: <sips:ngw1@a.example.com>
Content-Type: application/sdp
Content-Length: 146

v=0
o=GW 2890844526 2890844526 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 ngw1.a.example.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.a.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS ngw1.a.example.com:5061;branch=z9hG4bKwqwee65

```
     ;received=192.0.2.103
   Max-Forwards: 69
   Record-Route: <sips:ss1.a.example.com;lr>
   From: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sips:+19185553333@ss1.a.example.com;user=phone>
   Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sips:ngw1@a.example.com>
   Content-Type: application/sdp
   Content-Length: 146

   v=0
   o=GW 2890844526 2890844526 IN IP4 ngw1.a.example.com
   s=-
   c=IN IP4 ngw1.a.example.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.a.example.com:5061;branch=z9hG4bK2d4790.1
     ;received=192.0.2.111
   Via: SIP/2.0/TLS ngw1.a.example.com:5061;branch=z9hG4bKlueha2
     ;received=192.0.2.103
   Record-Route: <sips:ss1.a.example.com;lr>
   From: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sips:4443333@gw2.a.example.com>
```

```
Content-Type: application/sdp
Content-Length: 143

v=0
o=GW 987654321 987654321 IN IP4 gw2.a.example.com
s=-
c=IN IP4 gw2.a.example.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.a.example.com:5061;branch=z9hG4bKlueha2
 ;received=192.0.2.103
Record-Route: <sips:ss1.a.example.com;lr>
From: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
To: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.com
CSeq: 1 INVITE
Contact: <sips:4443333@gw2.a.example.com>
Content-Type: application/sdp
Content-Length: 143

v=0
o=GW 987654321 987654321 IN IP4 gw2.a.example.com
s=-
c=IN IP4 gw2.a.example.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.a.example.com:5061;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TLS ngw1.a.example.com:5061;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sips:ss1.a.example.com;lr>
   From: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sips:4443333@gw2.a.example.com>
   Content-Type: application/sdp
   Content-Length: 143

   v=0
   o=GW 987654321 987654321 IN IP4 gw2.a.example.com
   s=-
   c=IN IP4 gw2.a.example.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.a.example.com:5061;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Record-Route: <sips:ss1.a.example.com;lr>
   From: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sips:4443333@gw2.a.example.com>
   Content-Type: application/sdp
   Content-Length: 143

   v=0
   o=GW 987654321 987654321 IN IP4 gw2.a.example.com
   s=-
   c=IN IP4 gw2.a.example.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.a.example.com SIP/2.0
   Via: SIP/2.0/TLS ngw1.a.example.com:5061;branch=z9hG4bKlueha2
   Max-Forwards: 70
   Route: <sips:ss1.a.example.com;lr>
   From: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
   CSeq: 1 ACK
   Content-Length: 0
```


   F14 ACK Proxy 1 -> GW 2

```
   ACK sips:4443333@gw2.a.example.com SIP/2.0
   Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/TLS ngw1.a.example.com:5061;branch=z9hG4bKlueha2
    ;received=192.0.2.103
   Max-Forwards: 69
   From: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   To: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
   Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.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@a.example.com SIP/2.0
   Via: SIP/2.0/TLS gw2.a.example.com:5061;branch=z9hG4bKtexx6
   Max-Forwards: 70
   Route: <sips:ss1.a.example.com;lr>
   From: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
   To: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.com
   CSeq: 4 BYE
   Content-Length: 0



   F17 RLC GW 2 -> Switch C

   RLC



   F18 BYE Proxy 1 -> NGW 1


   BYE sips:ngw1@a.example.com SIP/2.0
   Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1
   Via: SIP/2.0/TLS gw2.a.example.com:5061;branch=z9hG4bKtexx6
    ;received=192.0.2.202
   Max-Forwards: 69
   From: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
   To: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.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.a.example.com:5061;branch=z9hG4bK2d4790.1
    ;received=192.0.2.111
   Via: SIP/2.0/TLS gw2.a.example.com:5061;branch=z9hG4bKtexx6
    ;received=192.0.2.202
   From: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
   To: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.com
   CSeq: 4 BYE
   Content-Length: 0
```
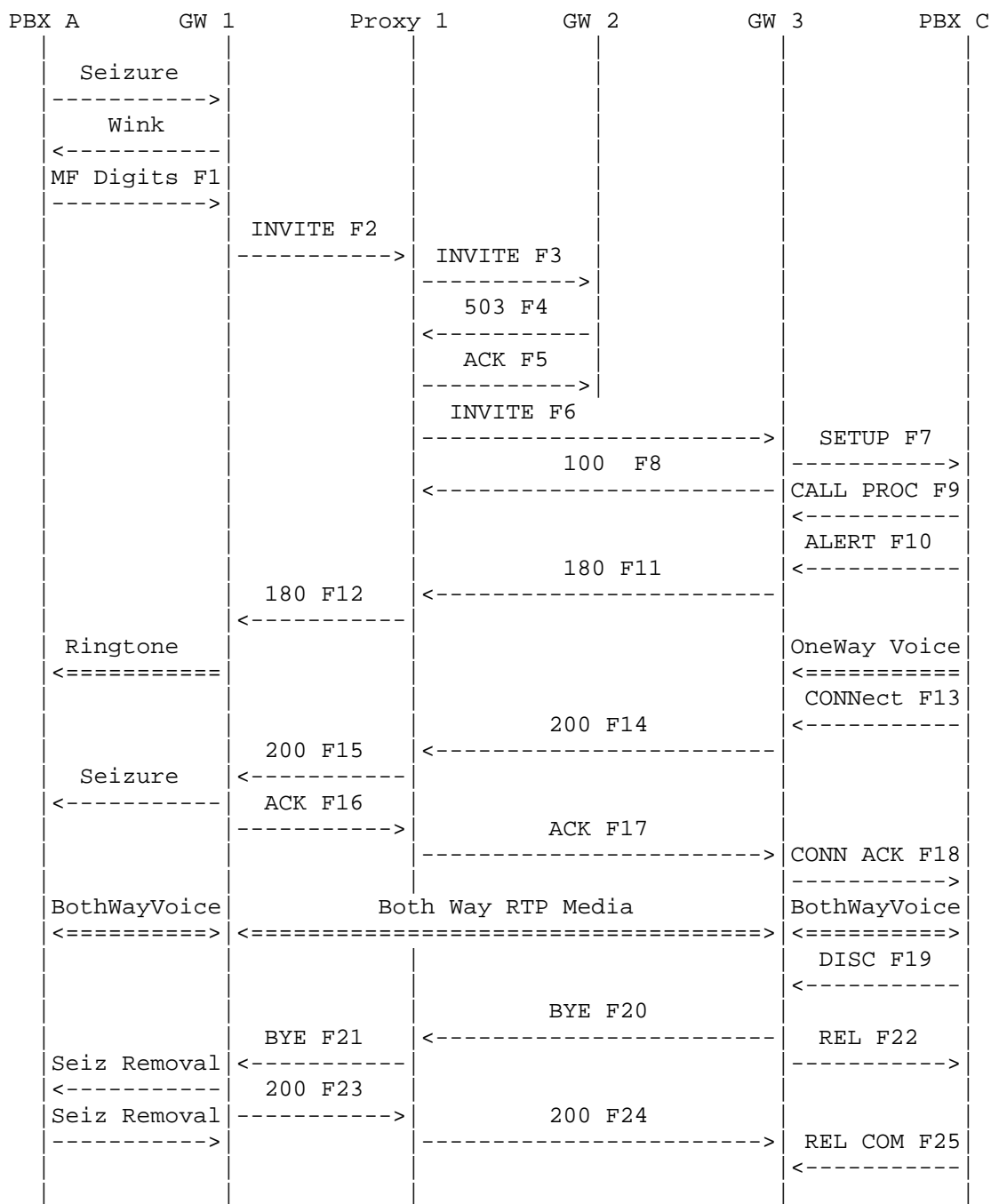
```
   F20 200 OK Proxy 1 -> GW 2

   SIP/2.0 200 OK
   Via: SIP/2.0/TLS gw2.a.example.com:5061;branch=z9hG4bKtexx6
    ;received=192.0.2.202
   From: <sips:+19185553333@ss1.a.example.com;user=phone>;tag=314159
   To: <sips:+13145551111@ngw1.a.example.com;user=phone>;tag=7643kals
   Call-ID: 2xTb9vxSit55XU7p8@ngw1.a.example.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.a.example.com SIP/2.0
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
   Max-Forwards: 70
   From: <sip:551313@gw1.a.example.com>;tag=63412s
   To: <sip:4443333@ss1.a.example.com>
   Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:551313@gw1.a.example.com>
   Content-Type: application/sdp
   Content-Length: 155

   v=0
   o=GW 2890844526 2890844526 IN IP4 gw1.a.example.com
   s=-
   c=IN IP4 gw1.a.example.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.a.example.com SIP/2.0
Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
Max-Forwards: 69
Record-Route: <sip:ss1.a.example.com;lr>
From: <sip:551313@gw1.a.example.com>;tag=63412s
To: <sip:4443333@ss1.a.example.com>
Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
CSeq: 1 INVITE
Contact: <sip:551313@gw1.a.example.com>
Content-Type: application/sdp
Content-Length: 155


v=0
o=GW 2890844526 2890844526 IN IP4 gw1.a.example.com
s=-
c=IN IP4 gw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.1
 ;received=192.0.2.111
Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
From: <sip:551313@gw1.a.example.com>;tag=63412s
To: <sip:4443333@ss1.a.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
CSeq: 1 INVITE
Content-Length: 0


F5 ACK Proxy 1 -> GW 2

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

```
Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
Max-Forward: 70
From: <sip:551313@gw1.a.example.com>;tag=63412s
To: <sip:4443333@ss1.a.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
CSeq: 1 ACK
Content-Length: 0
```


F6 INVITE Proxy 1 -> GW 3

```
INVITE sip:+19185553333@gw3.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.2
Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
Max-Forwards: 69
Record-Route: <sip:ss1.a.example.com;lr>
From: <sip:551313@gw1.a.example.com>;tag=63412s
To: <sip:4443333@ss1.a.example.com>
Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
CSeq: 1 INVITE
Contact: <sip:551313@gw1.a.example.com>
Content-Type: application/sdp
Content-Length: 155
```

```
v=0
o=GW 2890844526 2890844526 IN IP4 gw1.a.example.com
s=-
c=IN IP4 gw1.a.example.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.a.example.com:5060;branch=z9hG4bKwqwee65
     ;received=192.0.2.201
    From: <sip:551313@gw1.a.example.com>;tag=63412s
    To: <sip:4443333@ss1.a.example.com>
    Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.2
     ;received=192.0.2.111
    Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
     ;received=192.0.2.201
    Record-Route: <sip:ss1.a.example.com;lr>
    From: <sip:551313@gw1.a.example.com>;tag=63412s
    To: <sip:4443333@ss1.a.example.com>;tag=123456789
    Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
    CSeq: 1 INVITE
    Contact: <sip:+19185553333@gw3.a.example.com;user=phone>
    Content-Length: 0


    F12 180 Ringing Proxy 1 -> GW 1

    SIP/2.0 180 Ringing
    Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
```

```
   ;received=192.0.2.201
Record-Route: <sip:ss1.a.example.com;lr>
From: <sip:551313@gw1.a.example.com>;tag=63412s
To: <sip:4443333@ss1.a.example.com>;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
CSeq: 1 INVITE
Contact: <sip:+19185553333@gw3.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.2
 ;received=192.0.2.111
Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
Record-Route: <sip:ss1.a.example.com;lr>
From: <sip:551313@gw1.a.example.com>;tag=63412s
To: <sip:4443333@ss1.a.example.com>;tag=123456789
Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
CSeq: 1 INVITE
Contact: <sip:+19185553333@gw3.a.example.com;user=phone>
Content-Type: application/sdp
Content-Length: 143

v=0
o=GW 987654321 987654321 IN IP4 gw3.a.example.com
s=-
c=IN IP4 gw3.a.example.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.a.example.com:5060;branch=z9hG4bKwqwee65
 ;received=192.0.2.201
Record-Route: <sip:ss1.a.example.com;lr>
From: <sip:551313@gw1.a.example.com>;tag=63412s
```

```
   To: <sip:4443333@ss1.a.example.com>;tag=123456789
   Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
   CSeq: 1 INVITE
   Contact: <sip:+19185553333@gw3.a.example.com;user=phone>
   Content-Type: application/sdp
   Content-Length: 143

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


   F17 ACK Proxy 1 -> GW 3

   ACK sip:+19185553333@gw3.a.example.com;user=phone SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.2
   Via: SIP/2.0/UDP gw1.a.example.com:5060;branch=z9hG4bKwqwee65
    ;received=192.0.2.201
   Max-Forwards: 69
   From: <sip:551313@gw1.a.example.com>;tag=63412s
   To: <sip:4443333@ss1.a.example.com>;tag=123456789
   Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.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.a.example.com SIP/2.0
   Via: SIP/2.0/UDP gw3.a.example.com:5060;branch=z9hG4bKkdjuwq
   Max-Forwards: 70
   Route: <sip:ss1.a.example.com;lr>
   From: <sip:4443333@ss1.a.example.com>;tag=123456789
   To: <sip:551313@gw1.a.example.com>;tag=63412s
   Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.com
   CSeq: 1 BYE
   Content-Length: 0


   F21 BYE Proxy 1 -> GW 1

   BYE sip:551313@gw1.a.example.com SIP/2.0
   Via: SIP/2.0/UDP ss1.a.example.com:5060;branch=z9hG4bK2d4790.2
   Via: SIP/2.0/UDP gw3.a.example.com:5060;branch=z9hG4bKkdjuwq
    ;received=192.0.2.203
   Max-Forwards: 69
   From: <sip:4443333@ss1.a.example.com>;tag=123456789
   To: <sip:551313@gw1.a.example.com>;tag=63412s
   Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.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.a.example.com:5060;branch=z9hG4bK2d4790.2
     ;received=192.0.2.111
    Via: SIP/2.0/UDP gw3.a.example.com:5060;branch=z9hG4bKkdjuwq
     ;received=192.0.2.203
    From: <sip:4443333@ss1.a.example.com>;tag=123456789
    To: <sip:551313@gw1.a.example.com>;tag=63412s
    Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.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.a.example.com:5060;branch=z9hG4bKkdjuwq
     ;received=192.0.2.203
    From: <sip:4443333@ss1.a.example.com>;tag=123456789
    To: <sip:551313@gw1.a.example.com>;tag=63412s
    Call-ID: 2xTb9vxSit55XU7p8@gw1.a.example.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

5. 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 3398 [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 include 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 [7], 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.

6. References

6.1. Normative References

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6.2. Informative References

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