SIP Service Examples

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Abstract

This informational document gives examples of SIP (Session Initiation Protocol) services. This covers most features offered in so-called Centrex offerings from local exchange carriers and PBX (Private Branch Exchange) features. Most of the services shown in this document are implemented in the SIP User Agents, although some require the assistance of a SIP Proxy. Some require some extensions to SIP including third party call control (3pcc) extensions such as the REFER method. These features are not intended to be an exhaustive set, but rather show implementations of common features likely to be implemented on SIP IP Telephones in a business environment.
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1 Overview

This informational document provides call flows detailing a SIP implementation of the following traditional telephony services:

- Call Hold
- Music on Hold/Call Park
- Unattended Transfer
- Consultation Hold
- Unconditional Call Forwarding
- Attended Transfer
- No Answer Call Forwarding
- Busy Call Forwarding
- Single-Line Extension
- 3-way Call
- Incoming Call Screening
- Find-Me
- Call Pickup
- Outgoing Call Screening

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

These flows assume the functionality described in the SIP Call Flow Examples document [2], which explores basic behavior and PSTN internetworking. Some of the scenarios described herein make use of "SIP Call Control Transfer" [3].

These flows were prepared assuming a network of proxies, registrars, PSTN gateways, and other SIP servers that have a pre-established trust relationship with each other, secured through other means than SIP. User agents wishing to use the services in this network are required to authenticate themselves with an edge proxy using SIP Digest. To improve the clarity of this document, authentication of User Agents is not explicitly shown in all flows, except where authentication directly relates to the service example.

These flows use SIP as defined by RFC 2543 [4] with some noted updates for RFC 2543bis. Note that this document is informational, and nothing stated here should be taken as normative. RFC 2543 and the other referenced documents are definitive as far as protocol issues are concerned. Also, these flows do not represent the only way to implement these services - other approaches such as Back-to-Back User Agents (B2BUA) may be more appropriate in some circumstances.

Each call flow is presented with a textual description of the scenario, a message flow diagram showing the messages exchanged between separate network elements, and the detailed contents of each message shown in the diagram.
1.1 Legend for Message Flows

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

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

Messages with parenthesis around name represent optional control messages.

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

/* Comments. */

1.2 Document History

The first version of this document was the Internet-Draft "draft-sparks-sip-service-examples.txt" October 1999.

The next version was combined with the SIP Telephony Call Flows document into the "draft-ietf-sip-call-flows-00.txt" April 2000.

This version is based on Section 7 of that document with many of the examples extensively rewritten using the REFER method.

1.3 Changes since 00

- Modified all REFER flows to include 202 Accepted and NOTIFY result message.
- Removed "Telephony" from title of document.
- Added note on 2.9 on use of REFER to exit 3-way conference.
- Modified Music on Hold/Call Park flow (2.2).
- Modified order of messages in transfers (2.4 and 2.5) to more closely resemble PSTN experience.
- Added Call Pickup flow (2.13).
- Added editorial notes on usefulness of Replaces header.
2 IP Telephony Services Features Call Flows

These call flows show how a number of standard telephony features can be implemented using SIP. They are not meant to represent a complete set. Some calls make use of SIP Call Control Extensions[3].

2.1 Call Hold

```
<table>
<thead>
<tr>
<th>User A</th>
<th>Proxy</th>
<th>User B</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE F1</td>
<td>INVITE F2</td>
<td></td>
</tr>
<tr>
<td>(100 Trying) F3</td>
<td>(180 Ringing) F4</td>
<td></td>
</tr>
<tr>
<td>180 Ringing F5</td>
<td>200 OK F6</td>
<td></td>
</tr>
<tr>
<td>200 OK F7</td>
<td>ACK F8</td>
<td></td>
</tr>
<tr>
<td>ACK F9</td>
<td>Both way RTP Established</td>
<td></td>
</tr>
<tr>
<td>INVITE (c=0) F10</td>
<td>INVITE (c=0) F11</td>
<td></td>
</tr>
<tr>
<td>200 OK F12</td>
<td>200 OK F13</td>
<td></td>
</tr>
<tr>
<td>ACK F14</td>
<td>No RTP Sent!</td>
<td></td>
</tr>
<tr>
<td>ACK F15</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```

User A calls User B, User B places call on hold. User B then takes call off hold. Note that either party can take the other party off of hold. User A hangs up call.

Message Details

F1 INVITE A -> Proxy 1

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

v=0
c=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 INVITE Proxy 1 -> B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ssl.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103

t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F3 (100 Trying) Proxy 1 -> A

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

F4 180 Ringing B -> Proxy 1

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing Proxy 1 -> A

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

F6 200 OK B -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0

F7 200 OK Proxy 1 -> A

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

v=0
c=IN IP4 110.111.112.113
m=audio 3456 RTP/AVP 0

F8 ACK A -> Proxy 1

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

F9 ACK Proxy 1 -> B

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 1 ACK
Content-Length: 0

/* User B places User A on hold. */

F10 INVITE B -> Proxy 1

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 0.0.0.0
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F11 INVITE Proxy 1 -> A

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 0.0.0.0
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
F12 200 OK A -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 0.0.0.0
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F13 200 OK Proxy 1 -> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 0.0.0.0
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F14 ACK B -> Proxy

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserB@there.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 1 ACK
Content-Length: 0

F15 ACK Proxy 1 -> A

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 1 ACK
Content-Length: 0

/* User B takes the call off hold */

F16 INVITE B -> Proxy 1

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 2 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F17 INVITE Proxy 1 -> A

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ssl.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 2 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F18 200 OK A -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 2 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F19 200 OK Proxy 1 -> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 2 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...
v=0
c=IN IP4 100.101.102.103
s=Session SDP
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F20 ACK B -> Proxy 1

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 2 ACK
Content-Length: 0

F21 ACK Proxy 1 -> A

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 2 ACK
Content-Length: 0

/* RTP Media stream re-established. User A disconnects. */

F22 BYE A -> Proxy 1

BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@there.com>
From: BigGuy <sip:UserA@here.com>;tag=1234567
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 2 BYE
Content-Length: 0

F23 BYE Proxy 1 -> B

BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=1234567
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 2 BYE
Content-Length: 0

F24 200 OK B -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=1234567
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 2 BYE
Content-Length: 0

F25 200 OK Proxy 1 -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=1234567
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 2 BYE
Content-Length: 0
2.2 Music on Hold/Call Park

User A  User B  Music/Park Server

INVITE F1
------------->
180 Ringing F2
<------------
200 OK F3
<------------
ACK F4
------------->
RTP Media
<===========>

User B Parks Call

REFER Refer-To: A F5
------------->
202 F6
------------->
INVITE F7 Referred-By: B
<---------------
200 OK F8
<-----------
ACK F9
<------------
RTP Music
<==========>

BYE F10
---------->
NOTIFY F12
200 OK F11
<----------
200 OK F13
<-------->

User B picks up the call

INVITE F14
<--------
200 F15
<--------
ACK F16
<--------
RTP Media
<========>

BYE F17
------------>
200 OK F18
<-----------

No more RTP Music

In this example, User A calls User B. User B then parks the call at the Music/Park Server by sending a REFER to the Music/Park Server.
The server sends an INVITE to A which replaces the session between A and B. The call is accepted by A and causes A to send a BYE to B. Note that the Replaces header is needed in this INVITE. User B receives notification of the successful park, and also receives the Call-ID in the application/sip body of the NOTIFY response. When User B wishes to retrieve the call, a new INVITE is sent to A which replaces the session with the Music/Park Server. User A accepts the call and sends a BYE to the Music/Park Server. Note that the Replaces header is needed in this INVITE as well. If someone besides User B retrieves the call, then this becomes Call Park and Pickup.

Message Details.

F1 INVITE A \to B

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

v=0
c=o=2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing B \to A

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

F3 200 OK B \to A

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

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F4 ACK A -> B

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

/* User B REFERs Music Server to establish session with A which replaces the established session between A and B */

F5 REFER B -> Music Server

REFER sip:music@server.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: Music Server <sip:music@server.com>
Call-ID: 4802029847@there.com
CSeq: 1 REFER
Refer-To: <sip:UserA@here.com>
Referred-By: <sip:UserB@there.com>
Content-Length: 0

F6 202 Accepted Music Server -> B

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=56323
To: Music Server <sip:music@server.com>
Call-ID: 4802029847@there.com
CSeq: 1 REFER
Content-Length: 0

/* Music Server places call to User A to replace session between A and B */

F7 INVITE Music Server -> A

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP server.com:5060
From: <sip:music@server.com>
To: BigGuy <sip:UserA@here.com>
Call-ID: a5-75-34-12-76@server.com
CSeq: 1 INVITE
Referred-By: <sip:UserB@there.com>
Contact: <sip:music@server.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=MusicServer 2890844526 2890844526 IN IP4 music.server.com
s=Session SDP
c=IN IP4 50.60.70.80
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F8 200 OK A-> Music Server

SIP/2.0 200 OK
Via: SIP/2.0/UDP server.com:5060
From: <sip:music@server.com>
To: BigGuy <sip:UserA@here.com>;tag=098594
Call-ID: a5-75-34-12-76@server.com
CSeq: 1 INVITE
Contact: <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly
F9 ACK Music Server -> A

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP server.com:5060
From: <sip:music@server.com>
To: BigGuy <sip:UserA@here.com>;tag=098594
Call-ID: a5-75-34-12-76@server.com
CSeq: 1 ACK
Content-Length: 0

F10 BYE A -> B

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

F11 200 OK B -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 2 BYE
Content-Length: 0

/* Music Server reports success back to B by returning all the SIP
headers in 200 OK response */

F12 NOTIFY Music Server -> B

NOTIFY sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
To: LittleGuy <sip:UserB@there.com>
From: Music Server <sip:music@server.com>;tag=56323
Call-ID: 4802029847@there.com
CSeq: 1 NOTIFY
Event: refer
Content-Type: application/sip
Content-Length: ...

SIP/2.0 200 OK
Via: SIP/2.0/UDP server.com:5060
From: <sip:music@server.com>
To: BigGuy <sip:UserA@here.com>
Call-ID: a5-75-34-12-76@server.com
CSeq: 1 INVITE
Contact: <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

F13 200 OK B -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
To: LittleGuy <sip:UserB@there.com>
From: Music Server <sip:music@server.com>;tag=56323
Call-ID: 4802029847@there.com
CSeq: 1 NOTIFY
Content-Length: 0

/* User A is now parked at the Music Server */

/* User B picks up the call by sending an INVITE to A which replaces
the existing session with the Music/Park Server. Note that B knows
the Call-ID of the existing session from the NOTIFY response. */

F14 INVITE B -> A

INVITE sip:UserA@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: LittleGuy <sip:UserB@there.com>
To: BigGuy <sip:UserA@here.com>
Call-ID: 6485356@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F15 200 OK A -> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: LittleGuy <sip:UserB@there.com>
To: BigGuy <sip:UserA@here.com>
Call-ID: 6485356@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

F16 ACK B -> A

ACK sip:UserA@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: LittleGuy <sip:UserB@there.com>
To: BigGuy <sip:UserA@here.com>
Call-ID: 6485356@here.com
CSeq: 1 ACK
Content-Length: 0

/* A replaces the session to the Music Server with the new session
and generates a BYE to disconnect the Music Server */

F17 BYE A -> Music Server

BYE sip:music@server.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=098594
To: <sip:music@server.com>
Call-ID: a5-75-34-12-76@server.com
CSeq: 1 BYE
Content-Length: 0

F18 200 OK Music Server -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=098594
To: <sip:music@server.com>
Call-ID: a5-75-34-12-76@server.com
CSeq: 1 BYE
Content-Length: 0
### Consultation Hold

<table>
<thead>
<tr>
<th>User A</th>
<th>Proxy</th>
<th>User B</th>
<th>User C</th>
</tr>
</thead>
<tbody>
<tr>
<td><img src="" alt="Message Flow Diagram" /></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
User A calls user B. User B places call on hold. User B calls User C. User B then disconnects with C, then takes the call with User A off hold. The call ends with B hangs up.

Message Details

F1 INVITE A -> Proxy 1

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

v=0
c=Client 289084526 289084526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 INVITE Proxy 1 -> B

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F3(100 Trying) Proxy 1 -> A

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

F4 180 Ringing B -> Proxy 1

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
F5 180 Ringing Proxy 1 -> A

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

F6 200 OK B -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F7 200 OK Proxy 1 -> A

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

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F8 ACK A -> Proxy 1

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

F9 ACK Proxy 1 -> B

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* User B places User A on hold. */

F10 INVITE B -> Proxy 1

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=Unicast 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 0.0.0.0
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F11 INVITE Proxy 1 -> A

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 0.0.0.0
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F12 200 OK A -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 0.0.0.0
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F13 200 OK Proxy 1 -> B
SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 0.0.0.0
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F14 ACK B -> Proxy 1

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F15 ACK Proxy 1 -> A

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F16 INVITE B -> Proxy 1

INVITE sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>
Call-ID: 9876543210@there.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=IN IP4 110.111.112.113
m=audio 50170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F17 INVITE Proxy 1 -> C

INVITE sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserC@anywhere.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>
Call-ID: 9876543210@there.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=IN IP4 110.111.112.113
m=audio 50170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F18 (100 Trying) Proxy 1 -> B

SIP/2.0 100 Trying
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>
Call-ID: 9876543210@there.com
CSeq: 1 INVITE
Content-Length: 0

F19 180 Ringing C -> Proxy 1
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@here.com
CSeq: 1 INVITE
Content Length:0

F20 180 Ringing Proxy 1 -> B
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@there.com
CSeq: 1 INVITE
Content Length: 0

F21 200 OK C -> Proxy 1
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserC@anywhere.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@there.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserC@anywhere.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=IN IP4 120.121.122.123
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F22 200 OK Proxy 1 -> B
SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserC@anywhere.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@there.com
CSeq: 1 INVITE
Contact: OtherGuy <sip:UserC@anywhere.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserC 2890844527 2890844527 IN IP4 client.anywhere.com
s=Session SDP
c=IN IP4 120.121.122.123
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F23 ACK B -> Proxy 1

ACK sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserC@anywhere.com>
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@there.com
CSeq: 1 ACK
Content-Length: 0

F24 ACK Proxy 1 -> C

ACK sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@there.com
CSeq: 1 ACK
Content-Length: 0

F25 BYE B -> Proxy 1

BYE sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserC@anywhere.com>
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@there.com
CSeq: 2 BYE
Content-Length: 0
F26 BYE Proxy 1 -> C

BYE sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@there.com
CSeq: 2 BYE
Content-Length: 0

F27 200 OK C -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@there.com
CSeq: 2 BYE
Content-Length: 0

F28 200 OK Proxy 1 -> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=456654
Call-ID: 9876543210@there.com
CSeq: 2 BYE
Content-Length: 0

/* User B takes the call off hold */

F29 INVITE B -> Proxy 1

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F30 INVITE Proxy 1 -> A

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F31 200 OK A -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F32 200 OK Proxy 1 -> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
Record-Route: <sip:UserA@here.com;maddr=ss1.wcom.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=Client LittleGuy UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F33 ACK B -> Proxy 1

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 2 ACK
Content-Length: 0

F34 ACK Proxy 1 -> A

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 2 ACK
Content-Length: 0

F35 BYE A -> Proxy 1
BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB@there.com>
From: BigGuy <sip:UserA@here.com>;tag=1234567
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F36 BYE Proxy 1 -> B

BYE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=1234567
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F37 200 OK B -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=1234567
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F38 200 OK Proxy 1 -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=1234567
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
2.4 Unattended Transfer

User B call User A. User A then transfers User B to User C, then User A disconnects with User B. User B establishes the session to C then reports the success back to A in the NOTIFY. If the transfer fails, User B can send a new INVITE back to A to re-establish the session.
Message Details

F1 INVITE B \rightarrow A

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

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

F2 180 Ringing A \rightarrow B

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

F3 200 OK A \rightarrow B

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

v=0
c=IN IP4 100.101.102.103
m=audio 3456 RTP/AVP 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F4 ACK B -> A

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

/* Session is established between A and B. */

/* A performs unattended transfer of B to C */

F5 REFER A -> B

REFER sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 2 REFER
Refer-To: <sip:UserC@anywhere.com>
Referred-By: <UserA@here.com>
Content-Length: 0

F6 202 Accepted B -> A

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP there.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 2 REFER
Content-Length: 0

/* A now disconnects with B */

F7 BYE A -> B

BYE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 3 BYE
Content-Length: 0

F8 200 OK B -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345601@here.com
CSeq: 3 BYE
Content-Length: 0

/* B attempts the transfer to C */

F9 INVITE B -> C

INVITE sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>
Call-ID: 7436222@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB@there.com>
Referred-By: <UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844539 2890844539 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423821 0
m=audio 3458 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F10 180 Ringing C -> B

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=928287
Call-ID: 7436222@here.com
CSeq: 1 INVITE

F11 200 OK C -> B
SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=928287
Call-ID: 7436222@here.com
CSeq: 1 INVITE
Contact: OtherGuy <sip:UserC@anywhere.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=IN IP4 120.121.122.123
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F12 ACK B -> C

ACK sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=928287
Call-ID: 7436222@here.com
CSeq: 1 ACK
Content-Length: 0

/* B and C now have established a session. B reports success to A which A probably ignores. */

F13 NOTIFY B -> A

NOTIFY sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=314159
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345601@here.com
CSeq: 1 NOTIFY
Event: refer
Content-Type: application/sip
Content-Length: ...

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=928287
Call-ID: 7436222@here.com
CSeq: 1 INVITE
Contact: OtherGuy <sip:UserC@anywhere.com>
Content-Type: application/sdp
Content-Length: ...

F14 200 OK A -> B

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

<table>
<thead>
<tr>
<th>User A</th>
<th>User B</th>
<th>User C</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE F1</td>
<td></td>
<td>No RTP Sent!</td>
</tr>
<tr>
<td>(100 Trying) F2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>180 Ringing F3</td>
<td>200 OK F4</td>
<td></td>
</tr>
<tr>
<td>ACK F5</td>
<td>RTP</td>
<td></td>
</tr>
<tr>
<td>INVITE c=0 F6</td>
<td>200 OK F7</td>
<td></td>
</tr>
<tr>
<td>ACK F8</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

INVITE F9
(100 Trying) F10
180 Ringing F11
200 OK F12
ACK F13
RTP
REFER Refer-To: A F14
202 Accepted F15

INVITE Referred-By: B F16
200 OK F17
ACK F18
RTP
BYE F19
User A calls User B. User B puts User A on hold then calls User C to announce transfer. User B transfers User C to User A which replaces the session between A and B. B then disconnects session with A. C reports success of transfer to B who then disconnects with C.

Note: the Replaces header would be very useful in this example.

Message Details

F1 INVITE A -> B

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

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 (100 Trying B -> A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
F3 180 Ringing B -> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=23431
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F4 200 OK B -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=23431
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F5 ACK A -> B

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

/* User A and User B have established a session. User B puts User A on Hold */

F6 INVITE B -> A

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=23431
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345600@here.com
CSeq: 1024 INVITE
Contact: BigGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client.there.com
s=Session SDP
c=IN IP4 0.0.0.0
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=23431
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345600@here.com
CSeq: 1024 ACK
Content-Length: 0

/* User B calls User C */
F9 INVITE B -> C

INVITE sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>
Call-ID: sdjfdjfskdf@there.com
CSeq: 42 INVITE
Contact: BigGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=IN IP4 110.111.112.113
m=audio 3458 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F10 (100 Trying C -> B)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>
Call-ID: sdjfdjfskdf@there.com
CSeq: 42 INVITE
Content-Length: 0

F11 180 Ringing C -> B

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@there.com
CSeq: 42 INVITE
Content-Length: 0

F12 200 OK C -> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@there.com
CSeq: 42 INVITE
Contact: OtherGuy <sip:UserC@anywhere.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserC 2890844527 2890844527 IN IP4 client.anywhere.com
s=Session SDP
c=IN IP4 120.121.122.123
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F13 ACK B -> C

ACK sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@there.com
CSeq: 42 ACK
Content-Length: 0

/* User B Transfers User C to User A */

F14 REFER B -> C

REFER sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@there.com
CSeq: 43 REFER
Refer-To: <sip:UserB@here.com?Accept-Contact=sip:UserB@here.com;only=true>
Referred-By: <sip:UserB@there.com>
Content-Length: 0

F15 202 Accepted A -> B

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@there.com
CSeq: 43 REFER
Content-Length: 0
/* User C establishes session with User A which replaces the session between User A and User B */

F16 INVITE C -> A

INVITE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP anywhere.com:5060
From: OtherGuy <sip:UserC@anywhere.com>
To: BigGuy <sip:UserA@here.com>
Call-ID: 9435674543@anywhere.com
CSeq: 1 INVITE
 REFERRED-By: <sip:UserB@there.com>
Accept-Contact: sip:UserB@here.com;only=true
Contact: OtherGuy <sip:UserC@anywhere.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserC 2890844529 2890844529 IN IP4 client.anywhere.com
s=Session SDP
c=IN IP4 120.121.122.123
t=3034423643 0
m=audio 3458 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F17 200 OK A -> C

SIP/2.0 200 OK
Via: SIP/2.0/UDP anywhere.com:5060
From: OtherGuy <sip:UserC@anywhere.com>
To: BigGuy <sip:UserA@here.com>;tag=ff3a
Call-ID: 9435674543@anywhere.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844538 2890844538 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423452 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F18 ACK C -> A

ACK sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP anywhere.com:5060
From: BigGuy <sip:UserA@here.com>;tag=ff3a
To: LittleGuy <sip:UserB@there.com>
Call-ID: 123456000@here.com
CSeq: 2 BYE
Content-Length: 0

/* User C tells User B that the call has been successfully transferred */
F21 NOTIFY C -> B

NOTIFY sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
To:LittleGuy <sip:UserB@there.com>
From: OtherGuy <sip:UserC@anywhere.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@there.com
CSeq: 1 NOTIFY
Content-Type: application/sdp
Content-Length: ...

SIP/2.0 200 OK
Via: SIP/2.0/UDP anywhere.com:5060
From: OtherGuy <sip:UserC@anywhere.com>
To: BigGuy <sip:UserA@here.com>;tag=ff3a
Call-ID: 9435674543@anywhere.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

F22 200 OK B -> C

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
To:LittleGuy <sip:UserB@there.com>
From: OtherGuy <sip:UserC@anywhere.com>;tag=5f35a3
Call-ID: sdjfdjfskdf@there.com
CSeq: 1 NOTIFY
Content-Length: 0

/* User B disconnects with User C */

F23 BYE B -> C

BYE sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=5f35a3
Call-ID: 12345600@here.com
CSeq: 44 BYE
Content-Length: 0

F24 200 OK C -> B

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: LittleGuy <sip:UserB@there.com>
To: OtherGuy <sip:UserC@anywhere.com>;tag=5f35a3
Call-ID: 12345601@here.com
CSeq: 44 BYE
Content-Length: 0
2.6 Call Forwarding Unconditional

User A |
--------
| INVITE F1 |
| ---------------> INVITE F2 |
(100 Trying) F3 |
<--------------- 180 Ringing F4 |
180 Ringing F5 |
<--------------- 200 OK F6 |
200 OK F7 |
<---------------

ACK F8 |
<--------------- ACK F9 |
Both way RTP Established |
<==================================>

User B wants all calls forwarded to the PSTN. User A calls User B. The Proxy server rewrites the request URI, and forwards the INVITE to a Gateway. Details of messaging behind the Gateway are not shown.

Message Details

F1 INVITE A -> Proxy

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

v=0
c=UDP 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
/* Proxy forwards call by rewriting Request-URI */

F2 INVITE Proxy -> Gateway

INVITE sip:+19727293660@gw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F3 (100 Trying) Proxy -> A

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

F4 180 Ringing Gateway -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
F5 180 Ringing Proxy -> A

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

F6 200 OK Gateway -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:+19727293660@gw1.wcom.com;user=phone>
Content-Type: application/sdp
Content-Length: ...

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

F7 200 OK Proxy -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:+19727293660@gw1.wcom.com;user=phone>
Content-Type: application/sdp
Content-Length: ...

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

F8 ACK A -> Proxy

ACK sip:+19727293660@gw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:+19727293660@gw1.wcom.com;user=phone>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F9 ACK Proxy -> Gateway

ACK sip:+19727293660@gw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F10 BYE A -> Proxy 1

BYE sip:+19727293660@gw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:+19727293660@gw1.wcom.com;user=phone>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F11 BYE Proxy 1 -> Gateway

BYE sip:+19727293660@gw1.wcom.com;user=phone SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F12 200 OK Gateway -> Proxy 1

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314159
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F13 200 OK Proxy 1 -> A

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

User A     Proxy     User B1     User B2

INVITE F1 -------------> INVITE F2
(100 Trying) F3 <---------------->

486 Busy F4

ACK F5

INVITE F6

---------------> 180 Ringing F7

200 OK F9

--------------->

200 OK F10

--------------->

ACK F11

--------------->

ACK F12

Both way RTP Established

<================================================================================================>

180 Ringing F8

<---------------

200 OK F16

<---------------

User B wants calls to B1 forwarded to B2 if B1 is busy (this information is known to the proxy). User A calls B1, B1 is busy, the proxy server places call to B2.

Message Details

F1 INVITE A -> Proxy

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

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

F2 INVITE Proxy -> B1

INVITE sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: ...

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

F3 (100 Trying) Proxy -> A

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

F4 486 Busy Here B1 -> Proxy

SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F5 ACK Proxy -> B1

ACK sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* The proxy now forwards the call to B2 */

F6 INVITE Proxy -> B2

INVITE sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

F7 180 Ringing B2 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F8 180 Ringing Proxy -> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F9 200 OK B2 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB2@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client2.there.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F10 200 OK Proxy -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB2@there.com>
Content-Type: application/sdp
Content-Length: ...
v=0
o=UserB 2890844527 2890844527 IN IP4 client2.there.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F11 ACK A -> Proxy

ACK sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB2@there.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F12 ACK Proxy -> B2

ACK sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B2 */
/* User A eventually hangs up with User B2. */

F13 BYE A -> Proxy

BYE sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB2@there.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F14 BYE Proxy -> B2
BYE sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F15 200 OK B2 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F16 200 OK Proxy -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
### 2.8 Call Forwarding - No Answer

User B wants calls to B1 forwarded to B2 if B1 is not answered (information is known to the proxy server). User A calls B1 and no one answers. The proxy server then places the call to B2.

Message Details

F1 INVITE A -> Proxy
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

F2 INVITE Proxy -> B1

INVITE sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

F3 (100 Trying) Proxy -> A

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
F4 180 Ringing B1 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing Proxy -> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* B1 rings until a configurable timer expires in the Proxy. The Proxy sends Cancel and proceeds down the list of routes. */

F6 CANCEL Proxy -> B1

CANCEL sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

F7 200 OK B1 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

F8 487 Request Cancelled B1 -> Proxy
SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F9 ACK Proxy -> B1

ACK sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F10 INVITE Proxy -> B2

INVITE sip:UserB4@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

F11 180 Ringing B2 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F12 180 Proxy -> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F13 200 OK B2 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB2@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client2.there.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F14 200 OK Proxy -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB2@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=IN IP4 110.111.112.114
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F15 ACK A -> Proxy

ACK sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB2@there.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F16 ACK Proxy -> B2

ACK sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B2. User A Hangs Up with User B2. */

F17 BYE A -> Proxy

BYE sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB2@there.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F18 BYE Proxy -> B2
BYE sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F19 200 OK B2 -> Proxy
SIP/2.0 200 OK
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F20 200 OK Proxy -> A
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
2.9 3-way Conference

User A calls User B, User B then invites user C to a 3-way call. User B will mix the audio streams (act as the conference bridge). If user B drops out of the call then the entire call is dropped. This is not a fully meshed conference, and does not make use of the concepts in the call control draft.

The signaling for this scenario is as follows: User A calls User B, this establishes the call between A and B. User B calls User C, this establishes the call between B and C. User B will mix the audio streams, sending media originating at A to C, and media originating at C to A. There is no SIP signaling relationship between User A and User C.

The REFER method with the Replaces header could be used by User B to drop out of the call without disconnecting A and C.

2.10 Single Line Extension

Single Line Extension (Sequential, First Wins implementation), a call will ring several extensions in sequence. The extension to answer the call becomes the active set, no other sets may join the call.

The signaling is described in Section 2.11 of this document. It is anticipated that Single Line Extension will be associated with help desk/call center applications rather than individual users. The signaling for this implementation of Single Line Extension and Find-Me is the same, the difference may be in the provisioning of the service.

Note that the call flows for a Home Extension have not yet been designed.
2.11  
Find-Me

<table>
<thead>
<tr>
<th>User A</th>
<th>Proxy</th>
<th>User B1</th>
<th>User B2</th>
<th>User B3</th>
<th>User B4</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE F1</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
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<tr>
<td>--------------------</td>
<td>-------</td>
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<td>---------</td>
<td>---------</td>
<td>---------</td>
</tr>
<tr>
<td>(100 Trying) F3</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>180 Ringing F4</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>180 Ringing F5</td>
<td></td>
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</tr>
<tr>
<td>Timeout</td>
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<td></td>
</tr>
<tr>
<td>CANCEL F6</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F7</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>487 F8</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>ACK F9</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE F10</td>
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<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>480 Not Logged In F11</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>ACK F12</td>
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<td></td>
</tr>
<tr>
<td>INVITE F13</td>
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<td></td>
<td></td>
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<tr>
<td>486 Busy Here F14</td>
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<td></td>
<td></td>
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<tr>
<td>ACK F15</td>
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<td></td>
<td></td>
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<tr>
<td>INVITE F16</td>
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<tr>
<td>180 Ringing F17</td>
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</tr>
<tr>
<td>200 OK F19</td>
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<tr>
<td>ACK F21</td>
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<tr>
<td>ACK F22</td>
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<td></td>
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<tr>
<td>Both way RTP Established</td>
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<td></td>
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<td></td>
<td></td>
</tr>
<tr>
<td>BYE F23</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BYE F24</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
A call to a user will attempt to locate that user by calling locations from a list of contacts. The location to answer the call becomes the active set, no other sets may join the call.

It is anticipated that the Find-me feature will be associated with individual users. The signaling for the implementation of Single Line Extension and Find-Me is the same, the difference may be in the provisioning of the service.

Message Details

F1 INVITE A -> Proxy

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

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

F2 INVITE Proxy -> B1

INVITE sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com;branch=83749.1
Via: SIP/2.0/UDP here.com;5060
Record-Route: <sip:UserB@there.com;maddr=ssl.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

F3 (100 Trying) Proxy -> A

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

F4 180 Ringing B1 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing Proxy -> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* B1 rings for until a configurable timer in the Proxy expires. The Proxy then sends Cancel and proceeds down the list of routes. */

F6 CANCEL Proxy -> B1

CANCEL sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

F7 200 OK B1 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

F8 487 Request Cancelled B1 -> Proxy

SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F9 ACK Proxy -> B1

ACK sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F10 INVITE Proxy -> B2

INVITE sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

F11 480 Not Logged In B2 -> Proxy

SIP/2.0 480 Not Logged In
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314756
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F12 ACK Proxy -> B2

ACK sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=314756
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F13 INVITE Proxy -> B3

INVITE sip:UserB3@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.3
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...
v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F14 486 Busy Here B3 -> Proxy

SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.3
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F15 ACK Proxy -> B3

ACK sip:UserB3@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.3
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7654321
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F16 INVITE Proxy -> B4

INVITE sip:UserB4@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.4
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

F17 180 Ringing B4 -> Proxy
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.4
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7137136
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F18 180 Ringing B4 -> Proxy
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7137136
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F19 200 OK B4 -> Proxy
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.4
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7137136
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB4@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client4.there.com
s=Session SDP
c=IN IP4 110.111.112.116
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F20 200 OK Proxy -> A
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7137136
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB4@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client4.there.com
s=Session SDP
c=IN IP4 110.111.112.116
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F21 ACK A -> Proxy

ACK sip:UserB4@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <UserB4@there.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7137136
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F22 ACK Proxy -> B4

ACK sip:UserB4@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7137136
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B4*/

/* User B4 Hangs Up with User A. */

F23 BYE B4 -> Proxy

BYE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserA@here.com>
From: LittleGuy <sip:UserB@there.com>;tag=7137136
To: BigGuy <sip:UserA@here.com>
Call-ID: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0

F24 BYE Proxy -> A

BYE sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=7137136
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0

F25 200 OK A -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=7137136
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0

F26 200 OK Proxy -> B4

SIP/2.0 200 OK
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>;tag=7137136
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345600@here.com
CSeq: 1 BYE
Content-Length: 0
2.12 Call Management (Incoming Call Screening)

User A | Proxy | User B
-------|-------|-------
INVITE F1 | | 
305 Use Proxy F2 | | 
ACK F3 | | 
INVITE F4 | | 
407 Proxy Authorization F5 | | 
ACK F6 | | 
INVITE F7 | | 
403 Screening Failure (Terminating) F8 | | 
ACK F9 | | 

User B has an incoming call screening list, User A is included on the list of addresses User B will not accept calls from. User A attempts to call user B. Messages F1, F2, and F3 are included to show that User B does not accept INVITEs that have not been screened by the proxy.

Message Details

F1 INVITE A -> B

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

v=0
c=Client 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* User B only accepts INVITEs that have been screened by the proxy */

F2 305 Use Proxy B -> A
SIP/2.0 305 Use Proxy
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=342123
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: <sip:ss1.wcom.com>
Content-Length: 0

F3 ACK A -> B
ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=342123
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* A retries the call through the proxy */

F4 INVITE A -> Proxy 1
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

v=0
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
/* Proxy 1 challenges User A for authentication */

F5 407 Proxy Authorization Required Proxy 1 -> A

SIP/2.0 407 Proxy Authorization Required
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7886765
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Proxy-Authenticate: Digest realm="MCI WorldCom SIP",
    domain="sip:ss1.wcom.com", nonce="ea9c8e88df84f1c4341ae5e6c85a359",
    opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0

F6 ACK A -> Proxy 1

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=7886765
Call-ID: 12345600@here.com
CSeq: 2 ACK
Content-Length: 0

/* User A responds by sending an INVITE with authentication credentials in it. */

F7 INVITE A -> Proxy 1

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 3 INVITE
Contact: BigGuy <sip:UserA@here.com>
Proxy-Authorization: DIGEST username="UserA",
    realm="MCI WorldCom SIP", nonce="ae9137be1c87d175c2dd63302a0d6e0a",
    opaque="", uri="sip:ssl.wcom.com",
    response="bbaec39f943bdcb3620d90afc548a45c"
Content-Type: application/sdp
Content-Length: ...

v=0
c=IP4 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F8 403 Screening Failure (Terminating) Proxy 1 -> A

SIP/2.0 403 Screening Failure (Terminating)
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=ffe254
Call-ID: 12345600@here.com
CSeq: 3 INVITE
Content-Length: 0

F9 ACK A -> Proxy 1

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=ffe254
Call-ID: 12345600@here.com
CSeq: 3 ACK
Content-Length: 0
2.13
Call Management (Outgoing Call Screening)

User A has an outgoing call screening list, User B is included on the list of addresses User A will not be able to place a call to. User A attempts to call user B.

Message Details

F1 INVITE A -> Proxy 1

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

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Proxy 1 challenges User A for authentication */

F2 407 Proxy Authorization Required Proxy 1 -> A
SIP/2.0 407 Proxy Authorization Required
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=90210
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Proxy-Authenticate: Digest realm="MCI WorldCom SIP",
domain="sip:ss1.wcom.com", nonce="ea9c8e88df84f1ce4341ae6cbe5a359",
opaque="", stale=FALSE, algorithm=MD5
Content-Length: 0

F3 ACK A -> Proxy 1

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=90210
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* User A responds by sending an INVITE with authentication credentials in it. */

F4 INVITE A -> Proxy 1

INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Contact: BigGuy <sip:UserA@here.com>
Proxy-Authentication: DIGEST username="UserA", realm="MCI WorldCom SIP",
nonce="cb360afc54bbacec39f943bd820d9a45c", opaque="", uri="sip:ss1.wcom.com",
response="b9d2e5bcedc9f69ab2a9b44f270285a6"
Content-Type: application/sdp
Content-Length: ...

y=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=3034423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F5 403 Screening Failure (Originating) Proxy 1 -> A
SIP/2.0 403 Screening Failure (Originating)
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=18017
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Content-Length: 0

F6 ACK A -> Proxy 1

ACK sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=18017
Call-ID: 12345600@here.com
CSeq: 2 ACK
Content-Length: 0
2.14 Call Pickup

User A           Proxy          User B1             User B2

<table>
<thead>
<tr>
<th>INVITE F1</th>
<th>INVITE F2</th>
</tr>
</thead>
<tbody>
<tr>
<td>(100 Trying) F3</td>
<td>180 Ringing F4</td>
</tr>
<tr>
<td>180 Ringing F5</td>
<td>&lt;------------------</td>
</tr>
</tbody>
</table>

User B1 and B2 are part of a work group that can pick up each others calls. User A calls B1 who does not answer. User B2 wishes to pick up the call and sends a REGISTER with Expires:0 to the proxy which recognizes this as a call pickup not a register based on the Request-URI. The proxy server then forks the INVITE by sending it to B2 who answers. The proxy then cancels the B1 fork of the INVITE.

Message Details
F1 INVITE A -> Proxy

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

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

F2 INVITE Proxy -> B1

INVITE sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ssl.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

F3 (100 Trying) Proxy -> A

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

F4 180 Ringing B1 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F5 180 Ringing Proxy -> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* User B2 decides to pick up the call */

F6 REGISTER  B2 -> Proxy

REGISTER sip:pickupgroup37.ss1.wcom.com SIP/2.0
Via: SIP/2.0/UDP b2.there.com:5060
From: LittleGuy <sip:UserB2@there.com>
To: LittleGuy <sip:UserB2@there.com>
Call-ID: 563456212@b2.there.com
CSeq: 1 REGISTER
Contact: BigGuy <sip:UserB2@there.com>
Expires: 0
Content-Length: 0

F7 200 OK  Proxy -> B2

SIP/2.0 200 OK
Via: SIP/2.0/UDP b2.there.com:5060
From: LittleGuy <sip:UserB2@there.com>
To: LittleGuy <sip:UserB2@there.com>
Call-ID: 563456212@b2.there.com
CSeq: 1 REGISTER
Contact: BigGuy <sip:UserB2@there.com>
Expires: 0
Content-Length: 0

/* Proxy recognizes the REGISTER directed at pickgroup37.ssl.wcom.com as being a call pickup attempt for the pending INVITE to B1. The proxy then forks the INVITE and sends it to B2. */

INVITE sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: ...

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

Ringing B2 -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

Ringing Proxy -> A

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F11 200 OK B2 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB2@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client2.there.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F12 200 OK Proxy -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@there.com;maddr=ss1.wcom.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: LittleGuy <sip:UserB2@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844527 2890844527 IN IP4 client2.there.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* The Proxy detects the answer by B2 and cancels the B1 branch */

F13 CANCEL Proxy -> B1
CANCEL sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

F14 200 OK B1 -> Proxy
SIP/2.0 200 OK
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

F15 ACK A -> Proxy
ACK sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB2@there.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F16 ACK Proxy -> B2
ACK sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

F17 487 Request Cancelled B1 -> Proxy
SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP ssl.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

F18 ACK Proxy -> B1

ACK sip:UserB1@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B2. User A Hangs Up with User B2. */

F19 BYE A -> Proxy

BYE sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB2@there.com>
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F20 BYE Proxy -> B2

BYE sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F21 200 OK B2 -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss1.wcom.com:5060;branch=83749.1
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

F22 200 OK Proxy -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
Acknowledgements

The authors wish to thank the following individuals for their assistance and review of this call flows document: Rohan Mahey, Jonathan Rosenberg, Hemant Agrawal, Henry Sinnreich, Dean Willis, David Devanatham, Joe Pizzimenti, Matt Cannon, John Hearty, the whole MCI WorldCom IPOP Design team, Scott Orton, Greg Osterhout, Pat Sollee, Doug Weisenberg, Danny Mistry, Steve McKinnon, and Denise Ingram, Denise Caballero, Tom Redman, Ilya Slain, Pat Sollee, John Truetken, and others from MCI WorldCom, 3Com, Cisco, Lucent and Nortel.
3 References


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