SIP Telephony Service Examples

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Abstract

This document gives examples of SIP (Session Initiation Protocol) telephony 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 document provides call flows detailing a SIP implementation of the following traditional telephony services:

- Call Hold (with Music)
- Consultation Hold
- Unattended Transfer
- Attended Transfer
- Unconditional Call Forwarding
- Busy Call Forwarding
- No Answer Call Forwarding
- 3-way Call
- Single-Line Extension
- Find-Me
- Incoming Call Screening
- 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 "SIP Telephony Call Flow Examples" [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.

These flows use SIP as defined by RFC2543 [4].

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.

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.
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

User A           Proxy          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: 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 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: 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 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
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: 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
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
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 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: 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 1

ACK sip:UserA@here.com SIP/2.0
Via: SIP/2.0/UDP there.com:5060
Route: <sip:UserB@there.com>
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 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: 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 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: 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
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

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 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: 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 ss1.wcom.com:5060;branch=83749.1

F24 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
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
2.2 Music on Hold

+-------+-------+------------------+
| User A | User B | Music Server     |
|--------+-------+------------------|
|        |       | INVITE F1        |
|        | <-----| -----------------
|        |       | 100 Trying F2    |
|        | <-----| -----------------
|        |       | 180 Ringing F3   |
|        | <-----| -----------------
|        |       | 200 OK F4        |
In this example, User A calls User B. User B then places User A on hold, then REFERs User A to a Music Server for the music on hold service. User A agrees to the music on hold and sends an INVITE to the Music Server to establish the one way media session. Before User B takes User A off hold, a REFER is sent to tell User A to send a BYE to the Music Server to stop the music. Then User B takes User A off hold.

Message Details.

F1 INVITE A -> 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
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 there.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com
CSeq: 1 INVITE
Content-Length: 0

F3 180 Ringing B -> 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

F4 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: 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

F5 ACK A -> B
ACK 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: 1 ACK
Content-Length: 0

/* B places 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=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

F7 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 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

F8 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

/* User B REFERS User A to a Music On Hold Server */

F9 REFER B -> A

REFER 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: 2 REFER
Refer-To: <sip:music@server.com>
Content-Length: 0

F10 (100 Trying) A -> B

SIP/2.0 100 Trying
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: 2 REFER
Content-Length: 0

/* User A connects to Music On Hold Server */

F11 INVITE A -> Music Server

INVITE sip:music@server.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: <sip:music@server.com>
Call-ID: 12345601@here.com
CSeq: 2 INVITE
Contact: BigGuy <sip:UserA@here.com>
Referred-By: <sip:UserB@there.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

F12 200 OK Music Server -> A

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: <sip:music@server.com>;tag=se83kw1
Call-ID: 12345601@here.com
CSeq: 2 INVITE
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
a=sendonly

F13 ACK A -> Music Server

ACK sip:music@server.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: <sip:music@server.com>;tag=se83kw1
Call-ID: 12345601@here.com
CSeq: 2 ACK
Content-Length: 0

/* User A now has Music and reports success back to B */

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>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 2 REFER
Content-Length: 0

/* User B prepares to take A off hold by first stopping music */

F15 REFER B -> A
REFER 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>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 3 REFER
Refer-To: <sip:music@server.com?method=BYE>
Content-Length: 0

F16 (100 Trying) A -> B

SIP/2.0 100 Trying
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: 3 REFER
Content-Length: 0

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>
To: <sip:music@server.com>;tag=se83kw1
Call-ID: 12345601@here.com
CSeq: 3 BYE
Contact: BigGuy <sip:UserA@here.com>
Referred-By: <sip:UserB@there.com>
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>
To: <sip:music@server.com>;tag=se83kw1
Call-ID: 12345601@here.com
CSeq: 3 BYE
Content-Length: 0

/* Music is stopped, User A reports to B success */

F19 200 OK A -> B
SIP/2.0 200 OK
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: 3 REFER
Content-Length: 0

/* B now takes A off hold and continues session */

F20 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=314159
To: BigGuy <sip:UserA@here.com>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 4 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

F21 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>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 4 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

F22 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>;tag=1234567
Call-ID: 12345601@here.com
CSeq: 4 ACK
Content-Length: 0
2.3 Consultation Hold

User A | Proxy | User B | User C
-------|-------|-------|-------
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 | | | |
<===================================> | | | |
| | | | |
INVITE (c=0) F10 | | | |
| | | |
INVITE (c=0) F11 | | | |

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User A calls user B. User B places call on hold. User B calls User C, after that call is finished User B take the call with User A off hold.

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
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=BigGuy 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: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

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


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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 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: 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
o=UserB 2890844526 2890844526 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
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
o=UserB 2890844526 2890844526 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423619 0
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-Type: application/sdp
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
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
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


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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 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: 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
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=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 ss1.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 ss1.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

This example is taken directly from "SIP Call Control Transfer"[3] section 4.4.1.

```
User A          User B          User C
| INVITE F1      |                  |
|----------------->|
| 200 OK F2      |                  |
|----------------->|
| ACK F3         |                  |
|----------------->|
| RTP            |                  |
|<==============>|
| INVITE (hold) F4  |                |
|----------------->|
| 200 OK F5      |                  |
|----------------->|
| ACK F6         |                  |
|----------------->|
| No RTP Sent!   |                  |
|----------------->|
| REFER Refer-To: C F7  |
|----------------->|
| 100 Trying F8  |                  |
|<=================>|
| INVITE F9      |                  |
|----------------->|
| 200 OK F10     |                  |
|<----------------->|
```
User B call User A. User B then transfers User A to User C. User B hangs up the call.

### Message Details

**F1 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>
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 110.111.112.113
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```
F2 200 OK A -> 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
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 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

F4 INVITE A -> 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>;tag=314159  
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  
m=audio 49170 RTP/AVP 0  
a=rtpmap:0 PCMU/8000

F5 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: 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

F6 ACK A -> B

ACK 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: 1 ACK
Content-Length: 0

/* B is now on hold. A transfers B to C */

F7 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>
Content-Length: 0

F8 (100 Trying) B -> A

SIP/2.0 100 Trying
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

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: 12345601@here.com
CSeq: 2 INVITE
Contact: LittleGuy <sip:UserB@there.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 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: 12345601@here.com
CSeq: 2 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

F11 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: 12345601@here.com
CSeq: 2 ACK
Content-Length: 0

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

F12 200 OK B -> A

SIP/2.0 200 OK
/* A now disconnects with B */

F13 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

F14 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
2.5 Attended Transfer

```
<table>
<thead>
<tr>
<th>User A</th>
<th>User B</th>
<th>User C</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>INVITE F1</td>
<td></td>
</tr>
<tr>
<td></td>
<td>--------------&gt;</td>
<td></td>
</tr>
<tr>
<td>(100 Trying) F2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;----------------</td>
<td></td>
<td></td>
</tr>
<tr>
<td>180 Ringing F3</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;----------------</td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;----------------</td>
<td></td>
<td></td>
</tr>
<tr>
<td>ACK F5</td>
<td></td>
<td></td>
</tr>
<tr>
<td>--------------&gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>RTP</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;==========</td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE c=0 F6</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;----------------</td>
<td></td>
<td></td>
</tr>
<tr>
<td>200 OK F7</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;----------------</td>
<td></td>
<td></td>
</tr>
<tr>
<td>ACK F8</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```
User A calls User B. User B puts User A on hold, and calls User C to announce transfer. User B transfers User A to User C while maintaining sessions with both A and C. Once the session between A and C is successfully established, User B hangs up on both A and C.
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
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

F7 200 OK A -> B

SIP/2.0 200 OK
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: LittleGuy <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

F8 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=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 -> A

INVITE sip:UserA@here.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: 12345600@here.com
CSeq: 1025 INVITE
Contact: BigGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserB 2890844528 2890844528 IN IP4 client.there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=3034423645 0
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: 12345600@here.com
CSeq: 1025 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: 12345600@here.com
CSeq: 1025 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: 12345600@here.com
CSeq: 1025 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: 12345600@here.com
CSeq: 1025 ACK
Content-Length: 0

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

F14 REFER B -> A
REFER 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>;tag=23431
Call-ID: 12345600@here.com
CSeq: 1026 REFER
Refer-To: <sip:UserC@anywhere.com>
Content-Length: 0

F15 (100 Trying) A -> B

SIP/2.0 100 Trying
Via: SIP/2.0/UDP there.com:5060
From: LittleGuy <sip:UserB@there.com>
To: BigGuy <sip:UserA@here.com>;tag=23431
Call-ID: 12345600@here.com
CSeq: 1026 REFER
Content-Length: 0

/* User A calls User C */

F16 INVITE A -> C

INVITE sip:UserC@anywhere.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: OtherGuy <sip:UserC@anywhere.com>
Call-ID: 12345600@here.com
CSeq: 2 INVITE
Referred-By: <sip:UserB@there.com>
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
/* User A tells User B that the call has been successfully transferred */
F19 200 OK A -> 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=23431
Call-ID: 12345600@here.com
CSeq: 1026 REFER
Content-Length: 0

/* User B then disconnects from both User A and User C */

F20 BYE B -> A

BYE 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>;tag=23431
Call-ID: 12345600@here.com
CSeq: 1027 BYE
Content-Length: 0

F21 200 OK A -> 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=23431
Call-ID: 12345600@here.com
CSeq: 1027 BYE
Content-Length: 0

F22 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: 1028 BYE
Content-Length: 0

F23 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: 1028 BYE
Content-Length: 0
2.6 Call Forwarding Unconditional

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
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 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
da=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: LittleGuy <sip:7773660,phone-context=p1234@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
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
<table>
<thead>
<tr>
<th>INVITE F1</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>(100 Trying) F3</td>
<td>486 Busy F4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;--------------</td>
<td>&lt;----------------</td>
<td>----------------</td>
<td>----------------</td>
</tr>
<tr>
<td>180 Ringing F8</td>
<td>200 OK F9</td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;---------------</td>
<td>&lt;----------------</td>
<td>----------------</td>
<td>----------------</td>
</tr>
<tr>
<td>200 OK F10</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;---------------</td>
<td>----------------</td>
<td>----------------</td>
<td>----------------</td>
</tr>
<tr>
<td>ACK F11</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>---------------</td>
<td>----------------</td>
<td>----------------</td>
<td>----------------</td>
</tr>
<tr>
<td>200 OK F16</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;---------------</td>
<td>----------------</td>
<td>----------------</td>
<td>----------------</td>
</tr>
</tbody>
</table>
| Both way RTP Established
<p>|==================================================================|</p>
<table>
<thead>
<tr>
<th>BYE F13</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>200 OK F15</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>&lt;---------------</td>
<td>----------------</td>
<td>----------------</td>
<td>----------------</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
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
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 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
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
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

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

ACK Proxy -> B2
ACK sip:UserB2@there.com SIP/2.0
Via: SIP/2.0/UDP ssl1.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. */

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 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: 2 BYE
Content-Length: 0

F15 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=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

```
<table>
<thead>
<tr>
<th>User A</th>
<th>Proxy</th>
<th>User B1</th>
<th>User B2</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE F1</td>
<td></td>
<td>INVITE F2</td>
<td></td>
</tr>
<tr>
<td>(100 Trying) F3</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>180 Ringing F4</td>
<td>&lt;</td>
<td>200 OK F7</td>
<td></td>
</tr>
<tr>
<td>Request Timeout</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CANCEL F6</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>487 F8</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>ACK F9</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>INVITE F10</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```

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

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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:UserB2@there.com>;tag=123456


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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:UserB2@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:UserB2@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: ...

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

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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

F19 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

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.

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

IP: 180 F18 <---------- 180 Ringing F17

ACK F15 <------------------------

INVITE F16

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<table>
<thead>
<tr>
<th></th>
<th>INVITE F10</th>
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<th></th>
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</thead>
<tbody>
<tr>
<td></td>
<td>------------&gt;</td>
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<tr>
<td></td>
<td>480 Not Logged In F11</td>
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<td>ACK F12</td>
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<td>&lt;------------</td>
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<td>INVITE F13</td>
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<td>------------&gt;</td>
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<td></td>
<td>486 Busy Here F14</td>
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<td>&lt;------------</td>
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<tr>
<td></td>
<td>ACK F15</td>
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<td></td>
<td>&lt;------------</td>
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</tr>
<tr>
<td></td>
<td>INVITE F16</td>
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<td>------------&gt;</td>
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<tr>
<td></td>
<td>180 F18</td>
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<tr>
<td></td>
<td>200 OK F19</td>
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</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 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=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 &lt;sip:UserB@there.com&gt;
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 &lt;sip:UserA@here.com&gt;
To: LittleGuy &lt;sip:UserB@there.com&gt;;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 &lt;sip:UserA@here.com&gt;
To: LittleGuy &lt;sip:UserB@there.com&gt;;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: &lt;sip:UserB@there.com;maddr=ss1.wcom.com&gt;
From: BigGuy &lt;sip:UserA@here.com&gt;
To: LittleGuy &lt; sip:UserB@there.com &gt;
Call-ID: 12345600@here.com
CSeq: 1 INVITE

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Contact: BigGuy &lt; sip:UserA@here.com &gt;
Content-Type: application/sdp
Content-Length: ...

v=0
o=UnderTestA 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 \rightarrow 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 \rightarrow 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 \rightarrow 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: ...
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

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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
2.12 Call Management (Incoming Call Screening)
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.

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>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Proxy-Authenticate: Digest realm="MCI WorldCom SIP",
domain="wcom.com", nonce="ea9c8e88df84f1cece4341ae6cbe5a359",
opaque="", stale=FALSE, algorithm=MD5
Content-Length:0

F3 ACK A -> Proxy 1

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

/* User A responds 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-Authorization: DIGEST username="UserA", realm="MCI WorldCom
   SIP", nonce="ae9137be1c87d175c2dd63302a0d6e0a", opaque="",
   uri="sip:ss1.wcom.com", response="bbaec39f943bdc3b620d90af548a45c"
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=303423619 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F5 403 Screening Failure (Terminating) Proxy 1 \rightarrow A

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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: 2 INVITE
Content-Length:0

F6 ACK A \rightarrow 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: 2 ACK
Content-Length:0
2.13 Call Management (Outgoing Call Screening)

User A

<table>
<thead>
<tr>
<th>INVITE F1</th>
</tr>
</thead>
<tbody>
<tr>
<td>----------</td>
</tr>
<tr>
<td>407 Proxy Authorization F2</td>
</tr>
</tbody>
</table>

User B

<table>
<thead>
<tr>
<th>Ack F3</th>
</tr>
</thead>
</table>

Proxy

<table>
<thead>
<tr>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>403 Screening Failure (Originating) F5</td>
</tr>
</tbody>
</table>

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="wcom.com", nonce="ea9c8e88df84f1cecc4341ae6cbe5a359",
opaque="", stale=FALSE, algorithm=MD5
Content-Length:0
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. */

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-Authorization: DIGEST username="UserA", realm="MCI WorldCom SIP", nonce="cb360af654bbaec39f943bd820d9a45c", opaque="", uri="sip:UserB@there.com;maddr=ss1.wcom.com", response="b9d2e5bcd9c9f69ab2a9b44f270285a6"
Content-Type: application/sdp
Content-Length: ...

v=0
o=UserA 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
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
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