Session Initiation Protocol Exceptional Procedure Examples
draft-hasebe-sipping-exceptional-procedure-examples-02

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Abstract

This document gives examples of Session Initiation Protocol (SIP) exceptional-procedure call flows. These scenarios are confusing and this document shows the best practices to handle them. The elements in these call flows include SIP User Agents and Clients. The scenarios include SIP session establishment. Call flow diagrams and message details are shown.

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1. Overview

The call flows shown in this document were derived in the design of a SIP IP communications network. These examples are difficult to interpret the behaviors of user agents based on RFCs.

In various situations which may happen when SIP is implemented, especially when a situation which serves as a norm of implementing in RFC is not illustrated, by showing operation of a terminal or a server as an example, it will be a help to a SIP implementors.

For example, the sequence which CANCEL and 200 OK for INVITE cross each other is possible. INVITE transaction obviously exists from UAC’s point of view, when the UAC sends a CANCEL message. However, when the UAS sends a 200 OK response for INVITE and then receives CANCEL message, there is not INVITE transaction anymore from UAS’s point of view. In such a case, it’s not easy to specify the response from the UAS in RFCs.

This document clarifies SIP UA behaviors when messages cross each other as exceptional-procedure conditions.

By clarifying operation under exceptional-procedure conditions, different interpretations between implementations are avoided and interoperability is expected to be promoted.

It is the hope of the authors that this document will be useful for SIP implementors, designers, and protocol researchers and will help them achieve the goal of a standard implementation of RFC 3261 [1].
These call flows are based on the current version 2.0 of SIP in RFC 3261 [1] with SDP usage described in RFC 3264 [2].

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14, RFC 2119 [4].

1.1. General Assumptions

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

These flows do not assume specific underlying transport protocols such as TCP, TLS, and UDP. See the discussion in RFC 3261 for details on the transport issues for SIP.

1.2. Legend for Message Flows

Dashed lines (---) and slash lines (/\) represent signaling messages that are mandatory to the call scenario.(X) represents crossover of signaling messages. Arrow indicate the direction of message flow.

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

Messages with parentheses around their name represent optional messages.

Messages are identified in the Figures as F1, F2, etc. These numbers are used for references to the message details that follow the Figure. Comments in the message details are shown in the following form:

/* Comments. */

1.3. SIP Protocol Assumptions

This document does not prescribe the flows precisely as they are shown, but rather illustrates the principles for best practice. They are best practice usages (orderings, syntax, selection of features for the purpose, or handling of error) of SIP methods, headers and parameters. NOTE: The flows in this document must not be copied as they are by implementors because additional characteristics were incorporated into the document for ease of
explanation. To sum up, the procedures described in this document represent well-reviewed examples of SIP usage, which are best common practice according to IETF consensus.

For simplicity in reading and editing the document, there are a number of differences between some of the examples and actual SIP messages. Examples are: Call-IDs are often repeated; CSeq often begins, at 1; header fields are usually shown in the same order; usually only the minimum required header field set is shown; and and Accept, Allow, etc are not shown.

Actors:

<table>
<thead>
<tr>
<th>Element</th>
<th>Display Name</th>
<th>URI</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>User Agent</td>
<td>Alice</td>
<td>sip:<a href="mailto:alice@atlanta.example.com">alice@atlanta.example.com</a></td>
<td>192.0.2.101</td>
</tr>
<tr>
<td>User Agent</td>
<td>Bob</td>
<td>sip:<a href="mailto:bob@biloxi.example.com">bob@biloxi.example.com</a></td>
<td>192.0.2.201</td>
</tr>
<tr>
<td>User Agent</td>
<td>Carol</td>
<td>sip:<a href="mailto:carol@chicago.example.com">carol@chicago.example.com</a></td>
<td>192.0.2.202</td>
</tr>
<tr>
<td>Proxy Server</td>
<td></td>
<td>ss.atlanta.example.com</td>
<td>192.0.2.111</td>
</tr>
</tbody>
</table>

2. Exceptional Procedures

This section details exceptional procedures between two SIP User Agents (UAs): Alice and Bob. Alice (sip:alice@atlanta.example.com) and Bob (sip:bob@biloxi.example.com) are assumed to be SIP phones or SIP-enabled devices.

This document clarifies how SIP UA should behave when messages cross each other in exceptional-procedure conditions.

For example, the sequence which CANCEL and 200OK for INVITE cross each other is possible. INVITE transaction obviously exists from UAC’s point of view, when the UAC sends a CANCEL message. However, when the UAS sends a 200 OK response for INVITE and then receives CANCEL message, there is not INVITE transaction anymore from UAS’s point of view. Actually, the UAS state has already transit to confirmed dialog already.

Examples of such exceptional procedures are shown below.

2.1. CANCEL crossover

```
Alice                     Bob
<p>| |
|                        |</p>
<table>
<thead>
<tr>
<th>INVITE F1</th>
</tr>
</thead>
<tbody>
<tr>
<td>180 Ringing F2</td>
</tr>
</tbody>
</table>
```

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In this scenario, Alice sends a CANCEL and Bob sends a 200 OK response to the initial INVITE message at the same time. Then Bob sends a 481 response in response to the CANCEL from Alice. UAC can terminate the session by sending a BYE immediately after receiving 200 OK for INVITE. By transmitting a BYE after 200 OK, "it just means that the software in his phone needs to maintain state for a short while in order to clean up properly." (RFC3261,15)

In this sequence, it is recommended that caller terminates the session by sending a BYE.

Message Details

F1 INVITE Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
c=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

F3 CANCEL Alice -> Bob

CANCEL sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0

/* When Alice sends a CANCEL, INVITE transaction exists. */

F4 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 147
v=0
c=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=--
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Alice sends a CANCEL and Bob sends a 200 OK response to the initial INVITE message at the same time. From Bob’s point of view, an INVITE transaction is completed by sending of the final response (200 OK). A 200 OK and a CANCEL crossed each other and inconsistency has arisen in the state of INVITE transaction of Alice and Bob. */

F5 481 Call/Transaction Dose Not Exist Bob -> Alice

SIP/2.0 481 Call/Transaction Dose Not Exist
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 38482762982220188511@atlanta.example.com
CSeq: 1 CANCEL
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

/* The final response to INVITE transaction has already sent while CANCEL request targeting this INVITE transaction is received, so Bob returns a 481 response. */

F6 ACK Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 38482762982220188511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* Bob has sent the final response, and a CANCEL becomes invalid. RTP streams are established. */

F7 BYE Alice -> Bob
BYE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

F8 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

2.2. BYE crossover

<table>
<thead>
<tr>
<th>Alice</th>
<th>Bob</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>INVITE F1</td>
</tr>
<tr>
<td></td>
<td>-----------&gt;</td>
</tr>
<tr>
<td></td>
<td>180 Ringing F2</td>
</tr>
<tr>
<td></td>
<td>&lt;----------</td>
</tr>
<tr>
<td></td>
<td>200 OK F3</td>
</tr>
<tr>
<td></td>
<td>&lt;----------</td>
</tr>
<tr>
<td></td>
<td>ACK F4</td>
</tr>
<tr>
<td></td>
<td>&lt;----------</td>
</tr>
<tr>
<td></td>
<td>Both Way RTP Media</td>
</tr>
<tr>
<td></td>
<td>&lt;=============</td>
</tr>
<tr>
<td></td>
<td>BYE F5</td>
</tr>
<tr>
<td></td>
<td>----------</td>
</tr>
<tr>
<td></td>
<td>\ /</td>
</tr>
<tr>
<td></td>
<td>X</td>
</tr>
<tr>
<td></td>
<td>&lt;------</td>
</tr>
<tr>
<td></td>
<td>481 F8</td>
</tr>
<tr>
<td></td>
<td>----------</td>
</tr>
<tr>
<td></td>
<td>\ /</td>
</tr>
<tr>
<td></td>
<td>X</td>
</tr>
</tbody>
</table>
In this scenario, Alice and Bob send a BYE at the same time. A session is ended shortly after a BYE request is passed to a client transaction. According to 15.1.1 of RFC3261, a dialog seems to be completed by a response or timeout of a BYE. Therefore, UA can normally transmit and receive a request until it receives a response. However, when UA sends a BYE, it determines that the dialog is completed. So, in this scenario, it is recommended that UA ends a dialog immediately after sending a BYE. In this case, both UA return a 481 response.

Message Details

F1 INVITE Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE

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Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

F3 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 38482762982201885511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 147

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F4 ACK Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 38482762982201885511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between Alice and Bob */

/* Bob hangs up with Alice. Note that the CSeq is NOT 2, since
Alice and Bob maintain their own independent CSeq counts.
(The INVITE was request 1 generated by Alice, and the BYE is
request 1 generated by Bob) */

F5 BYE Alice -> Bob

BYE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

/* Alice terminates a session by sending a BYE.
RTP streams are terminated. */

/* The session is terminated by sending a BYE. Although the dialog
is completed by receiving a response or a timeout, when UA sends
a BYE, it determines that the dialog is completed. */

F6 BYE Bob -> Alice

BYE sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0

/* Bob has also transmitted a BYE simultaneously with Alice.
Bob terminates a session and dialog. */

F7 481 Call/Transaction Does Not Exist Bob -> Alice

SIP/2.0 481 Call/Transaction Does Not Exist
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds8
;received=192.0.2.201
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

/* Since the dialog is already terminated, the BYE is returned
by a 481. */

Editor’s Note:
UA would send a 200 OK response to the BYE or not?
It is thought that it is necessary to clarify the dialog state.

F8 481 Call/Transaction Does Not Exist Alice -> Bob
SIP/2.0 481 Call/Transaction Does Not Exist
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
 ;received=192.0.2.201
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76s1
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 BYE
Content-Length: 0

/* Since Bob has terminated the dialog by sending a BYE,
a BYE which Alice sent is also returned by a 481. */

2.3. Session timer crossover(re-INVITE,BYE)

In this scenario, Bob sends a re-INVITE, and Alice sends a BYE at the same time. The re-INVITE of Bob is returned by a 481. Although TU of Bob has terminated the dialog by sending a BYE,
the client transaction of a re-INVITE still exists. Therefore, a client transaction sends ACK to 481 response.

Message Details

F1 INVITE Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Supported: timer
Session-Expires: 300
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

F3 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
Require: timer
Session-Expires: 300;refresher=uas
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 147

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Since there was no specification of refresher, Bob sets refresher=uas. */

F4 ACK Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between Alice and Bob */

/* Bob hangs up with Alice. Note that the CSeq is NOT 2, since Alice and Bob maintain their own independent CSeq counts. (The INVITE was request 1 generated by Alice, and the BYE is request 1 generated by Bob) */

F5 BYE Alice -> Bob

BYE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
/* Alice sends a BYE and terminates a session and dialog. */

F6 re-INVITE Bob -> Alice

INVITE sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
Session-Expires: 300;refresher=uac
Max-Forwards: 70
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

/* Alice sends a BYE, and Bob sends a re-INVITE at the same time. From Alice’s point of view, the dialog is completed, and in Bob’s view, the dialog is terminated. The state of a dialog is mismatching. */

F7 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds8
;received=192.0.2.201
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

F8 481 Call/Transaction Does Not Exist Alice -> Bob

SIP/2.0 481 Call/Transaction Does Not Exist
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
;received=192.0.2.201
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

/* Since Alice has already terminated the dialog by sending a BYE, she returns a 481. */

F9 ACK Bob -> Alice
2.4. REFER crossover (REFER, BYE)

In this scenario, Bob sends a REFER, and Alice sends a BYE at the same time. Bob send a REFER in the same dialog. From Alice’s point of view, as 2.2 described, a dialog is terminated by sending a BYE request, and Alice returns a 481 to the REFER. (If a dialog is terminated after receiving the response of a BYE or expiration of timer, Alice returns 202 to the REFER and the REFER method is successful. Also when a dialog is terminated, it is not clear whether UA continues call transfer.)
Message Details

F1 INVITE Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Supported: timer
Session-Expires: 300
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
c=IN IP4 192.0.2.101
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

F3 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
Require: timer
Session-Expires: 300;refresher=uas
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 147

v=0
c=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F4 ACK Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between Alice and Bob */
/* Bob hangs up with Alice. Note that the CSeq is NOT 2, since Alice and Bob maintain their own independent CSeq counts.
(The INVITE was request 1 generated by Alice, and the BYE is request 1 generated by Bob) */

F5 BYE Alice -> Bob

BYE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

/* Alice sends a BYE and terminates a session and dialog. */

F6 REFER Bob -> Alice
REFER sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
Refer-To: sip:carol@cleveland.example.org
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
CSeq: 1 REFER
Content-Length: 0

/* Alice sends a BYE, and Bob sends a REFER at the same time. 
Bob sends a REFER as REFER method in the same dialog.
From Alice’s point of view, the dialog is completed, and in Bob’s view,
the dialog is terminated. The state of a dialog is mismatching. */

F7 200 OK Bob -> Alice
SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds8
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

F8 481 Call/Transaction Does Not Exist Alice -> Bob
SIP/2.0 481 Call/Transaction Does Not Exist
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
;received=192.0.2.201
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 REFER
Content-Length: 0

/* Since Alice has already terminated the dialog by sending a BYE,
she returns a 481. */

2.5. A BYE is sent immediately after sending of a re-INVITE

Alice                         Bob
<p>| |
|                             |</p>
<table>
<thead>
<tr>
<th>INVITE F1</th>
</tr>
</thead>
</table>

Hasebe [Page 19]
In this scenario, Bob sends a BYE immediately after sending a re-INVITE,
(A user is not conscious that refresher sends a re-INVITE automatically. For example, in the case of a telephone application, it is possible that a user places a receiver immediately after refresher.)
When Alice receives a BYE other than ACK, she stops retransmitting of 200 OK. Since ACK for 2xx responses is not a server transaction, it is that UAS core transmits directly. this case is different from the case of an error response in 2.4. With UAS core, since the dialog which matches 200 OK received is terminated, 200 OK is ignored, without sending ACK.

Message Details

F1 INVITE Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Supported: timer
Session-Expires: 300
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
c=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

F3 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
Require: timer
Session-Expires: 300;refresher=uas
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 147

v=0
c=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
F4 ACK Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between Alice and Bob */

/* Bob hangs up with Alice. Note that the CSeq is NOT 2, since Alice and Bob maintain their own independent CSeq counts. (The INVITE was request 1 generated by Alice, and the BYE is request 1 generated by Bob) */

F5 re-INVITE Bob -> Alice

INVITE sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
Session-Expires: 300;refresher=uac
Max-Forwards: 70
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

F6 BYE Bob -> Alice

BYE sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

/* Bob sends a BYE immediately after sending of a re-INVITE, Bob terminates a session and dialog, without receiving the response to re-INVITE. */

F7 200 OK Alice -> Bob
SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds7;received=192.0.2.201
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

/* Bob sends a BYE, and Alice returns 200 OK to a re-INVITE. The state of a dialog is mismatching. */

F8 200 OK Alice -> Bob

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds8;received=192.0.2.201
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

/* The UAC core of Bob does not send a ACK after receiving 200 OK to a re-INVITE. (Bob has terminated the dialog by sending of a BYE.) The UAS core of Alice does not retransmit 200 OK to a re-INVITE. (Since the dialog is terminated by reception of BYE, Alice does not retransmit 200 OK even if she does not receive ACK from Bob.) */

2.6. re-INVITE crossover

<table>
<thead>
<tr>
<th>Alice</th>
<th>Bob</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE F1</td>
<td></td>
</tr>
<tr>
<td>180 Ringing F2</td>
<td></td>
</tr>
<tr>
<td>200 OK F3</td>
<td></td>
</tr>
<tr>
<td>ACK F4</td>
<td></td>
</tr>
<tr>
<td>Both Way RTP Media</td>
<td></td>
</tr>
<tr>
<td>re-INVITE F5</td>
<td>re-INVITE F6</td>
</tr>
</tbody>
</table>
In this scenario, Alice and Bob send a re-INVITE at the same time. When two re-INVITEs cross in the same dialog, they resend re-INVITEs after different intervals. ([RFC3261], 14.1)

When Alice sends an initial INVITE, an INVITE will be sent again after 2.1-4.0 seconds because she generated the Call-ID (owner of the Call-ID). Bob will send an INVITE again after 0.0-2.0 seconds, because Bob isn’t the owner of the Call-ID. Therefore, each user agent must remember whether they has generated the Call-ID of the dialog or not, in case INVITEs may be crossed by another INVITE.
Message Details

F1 INVITE Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Supported: timer
Session-Expires: 300
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

F3 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
Require: timer
Session-Expires: 300;refresher=uas
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 147

v=0
c=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F4 ACK Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between Alice and Bob */

F5 re-INVITE Alice -> Bob

INVITE sip:sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Content-Length: 151

v=0
c=alice 2890844526 2890844527 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly

F6 re-INVITE Bob -> Alice
INVITE sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
Session-Expires: 300;refresher=uac
Max-Forwards: 70
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

/* A case where a re-INVITE for a session refresh and a re-INVITE for hold are sent at the same time. */

F7 491 Request Pending Bob -> Alice

SIP/2.0 491 Request Pending
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Content-Length: 0

/* Since an INVITE is in process, a 491 response are returned. */

F8 491 Request Pending Alice -> Bob

SIP/2.0 491 Request Pending
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

F9 ACK(INVITE) Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 ACK
Content-Length: 0

F10 ACK(INVITE) Alice -> Bob
ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

F11 re-INVITE Bob -> Alice
INVITE sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7.1
Session-Expires: 300;refresher=uac
Max-Forwards: 70
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Content-Type: application/sdp
Content-Length: 147

v=0
c=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Since Bob is not the owner of Call-ID, Bob sends an INVITE again after 0.0-2.0 seconds. */

F12 200 OK Alice -> Bob
SIP/2.0 200 OK
Via: SIP/2.0/UDP client.biloxi.example.com:5060;branch=z9hG4bKnashds7.1
Session-Expires: 300;refresher=uac
Max-Forwards: 70
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 INVITE
Content-Type: application/sdp
Content-Length: 151

v=0
c=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com

Hasebe
s=
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F13 ACK Bob -> Alice

ACK sip:alice@client.atlanta.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74b44
Session-Expires: 300;refresher=uac
Max-Forwards: 70
From: Bob <sip:bob@biloxi.example.com>;tag=8321234356
To: Alice <sip:alice@atlanta.example.com>;tag=9fxcxed76sl
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 ACK
Content-Length: 0

F14 re-INVITE Alice -> Bob

INVITE sip:sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9.1
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxcxed76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 3 INVITE
Content-Length: 151

v=0
o=alice 2890844526 2890844527 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendonly

/* Since Alice is the owner of Call-ID, Alice sends an INVITE again after 2.1-4.0 seconds. */

F15 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9.1
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxcxed76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 3 INVITE
Content-Length: 151

v=0
c=bob 2890844527 2890844528 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=recvonly

F16 ACK Alice -> Bob

ACK sip:sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK230f2.1
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 3 ACK
Content-Length: 0

2.7. Initial INVITE retransmission

In this scenario, all provisional responses to the initial INVITE (ini-INVITE) are lost, and UAC retransmits an ini-INVITE. At the same time as retransmission, UAS generates a 200 OK to the ini-INVITE and it terminate an INVITE server transaction.
(RFC3261, 13.3.1.4) After sending a 200 OK, a TU of UAS processes ACK and retransmission of 200 OK. (RFC3261, 17.1) A TU of UAS processes a retransmitted ini-INVITE, but it doesn’t have a To-tag, therefore the TU cannot use a mechanism to recognize the dialog by From-tag, Call-ID and To-tag. However, it must recognize a retransmitted ini-INVITE correctly by From-tag and Call-ID. (It must not construct a new dialog in response to a retransmitted ini-INVITE regarding it as a request outside dialog.)

Since TU of UAS retransmits a 200 OK according to the timer during it waits for ACK, it doesn’t need to retransmit 200 OK for the retransmitted ini-INVITE.

Message Details

F1 INVITE Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
c=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

/* A 180 response is lost and does not reach Alice. */
F3 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 147

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Bob sends a 200 OK and terminates the INVITE transaction at the same time. TU performs retransmitting of a 200 OK directly.*/

F4 INVITE(retransmission) Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* Since F2 is lost, Alice retransmits an ini-INVITE. The retransmitted INVITE does not match any existing transaction because Bob has
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already sent a 200 OK. Moreover, the retransmitted ini-INVITE does
doesn’t match any existing dialog.
Therefore, Bob have to recognize the retransmitted INVITE correctly,
without treating with the new INVITE. */

F5 ACK Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

2.8. Early Dialog

<table>
<thead>
<tr>
<th>Alice</th>
<th>Proxy</th>
<th>Bob</th>
<th>Carol</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE(sdp1) F1</td>
<td>-------------&gt; INVITE(sdp1) F2</td>
<td></td>
<td></td>
</tr>
<tr>
<td>100 Trying F3</td>
<td>-------------&gt; 183(sdp2)To-tag=1 F4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>183(sdp2)To-tag=1 F5</td>
<td>&lt;------------</td>
<td>PRACK F6</td>
<td></td>
</tr>
<tr>
<td>PRACK F6</td>
<td>&lt;------------</td>
<td>-------------&gt; PRACK F7</td>
<td></td>
</tr>
<tr>
<td>200(PRACK) F9</td>
<td>&lt;------------</td>
<td></td>
<td>200(PRACK) F8</td>
</tr>
<tr>
<td>&lt;&lt;&lt;Both Way RTP Media</td>
<td>-----</td>
<td></td>
<td></td>
</tr>
<tr>
<td>CANCEL F10</td>
<td>-----</td>
<td>200(CANCEL) F11</td>
<td></td>
</tr>
<tr>
<td></td>
<td>-----</td>
<td>487(INVITE) F12</td>
<td></td>
</tr>
<tr>
<td></td>
<td>-----</td>
<td>ACK(INVITE) F13</td>
<td></td>
</tr>
<tr>
<td></td>
<td>-----</td>
<td>INVITE(sdp1) F14</td>
<td></td>
</tr>
<tr>
<td></td>
<td>-----</td>
<td>180 To-tag=2 F16</td>
<td>&lt;------------</td>
</tr>
</tbody>
</table>
In this scenario, a proxy is forking to another address (Carol), when Bob don’t return final response. When a proxy sends INVITEs to two or more addresses, two or more early dialogs may be established at UAC. At UAC, all early dialogs are released when the final response of ini-INVITE is received. (RFC3261, 13.2.2.4)

Only the confirmed dialog continues after a 200 OK reception. Even if Bob is replaced by Media Server, you have the result appear to UAC just as this call flow. In this sequence, Bob can terminate the early media when he receives CANCEL, but Alice does not have the trigger to terminate the early dialog. When Bob stops sending RTP by the CANCEL from Proxy, it seems to Alice that RTP breaks off suddenly.

Message Details

F1 INVITE Alice -> Proxy

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Supported: 100rel
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
F2 INVITE Proxy -> Bob

INVITE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Supported: 100rel
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F3 100 Trying Proxy -> Alice

SIP/2.0 100 Trying
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

F4 183 Session Progress Bob -> Proxy

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.1
 ;received=192.0.2.233
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Require: 100rel
Content-Type: application/sdp
Content-Length: 148

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

F5 183 Session Progress Proxy -> Alice

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
RSeq: 1
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Require: 100rel
Content-Type: application/sdp
Content-Length: 148

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

/* Early dialog is established between Alice and Bob, and early media is also established at the same time. */

F6 PRACK Alice -> Proxy

PRACK sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 2 PRACK
RAck: 1 1 INVITE
Content-Length: 0

F7 PRACK Proxy -> Bob

PRACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.2
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 2 PRACK
RAck: 1 1 INVITE
Content-Length: 0

F8 200 OK(PRACK) Bob -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.2
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 2 PRACK
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

F9 200 OK(PRACK) Proxy -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 2 PRACK
Content-Length: 0

F10 CANCEL Proxy -> Bob

CANCEL sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.2
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 CANCEL
/* The proxy cancels the INVITE to Bob, and sends an INVITE to Carol. Although the continuation of early media after CANCEL reception depends on the implementation of the user agent, Alice does not know that the proxy cancels the INVITE, therefore, when Bob stops early media after CANCEL reception, it seems to her that the sound stops suddenly. */

F11 200 OK(CANCEL) Bob -> Proxy
SIP/2.0 200 OK
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.2
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0

F12 487 Request Terminated(INVITE) Bob -> Proxy
SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

F13 ACK(INVITE) Proxy -> Bob
ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.1
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

F14 INVITE Proxy -> Carol
INVITE sip:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK83749a.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
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; received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Supported: 100rel
Content-Type: application/sdp
Content-Length: 151

v=0
c=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F15 180 Ringing Carol -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK83749a.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
; received=192.0.2.101
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=456654
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:carol@client.chicago.example.com>

F16 180 Ringing Proxy -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=456654
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:carol@client.chicago.example.com>
Content-Length: 0

/* Proxy and Alice establish the second Early dialog when they
receive a 180 response from Carol. */

F17 200 OK Carol -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK83749a.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=456654
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:carol@client.chicago.example.com>
Content Length: 151

v=0
o=carol 2890844922 2890844922 IN IP4 client.chicago.example.com
s=Session SDP
c=IN IP4 client.chicago.example.com
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F18 200 OK Proxy -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
 ;received=192.0.2.101
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=456654
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:carol@client.chicago.example.com>
Content Length: 151

v=0
o=carol 2890844922 2890844922 IN IP4 client.chicago.example.com
s=Session SDP
c=IN IP4 client.chicago.example.com
t=3034423619 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* By 200 OK response, all early dialogs are terminated except
for ones that was confirmed. */
F19 ACK Alice -> Proxy

ACK sip:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bq9
Max-Forwards: 70
Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=456654
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

F20 ACK Proxy -> Carol

ACK sip:carol@client.chicago.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bq9
 ;received=192.0.2.101
Max-Forwards: 69
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=456654
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

2.9. CANCEL crossover via a stateful proxy

<table>
<thead>
<tr>
<th>Alice</th>
<th>Proxy</th>
<th>Bob</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>INVITE F1</td>
<td></td>
</tr>
<tr>
<td></td>
<td>100 Trying F3</td>
<td>INVITE F2</td>
</tr>
<tr>
<td></td>
<td>180 F5</td>
<td>180 Ringing F4</td>
</tr>
<tr>
<td></td>
<td>CANCEL F6</td>
<td>200 F8</td>
</tr>
<tr>
<td></td>
<td>200(CANCEL) F9</td>
<td>CANCEL F7</td>
</tr>
<tr>
<td></td>
<td>\ /</td>
<td>200 F8</td>
</tr>
<tr>
<td></td>
<td>X</td>
<td></td>
</tr>
<tr>
<td></td>
<td>/ \</td>
<td></td>
</tr>
<tr>
<td></td>
<td>200(INVITE) F10</td>
<td>481(CANCEL) F11</td>
</tr>
<tr>
<td></td>
<td>ACK F12</td>
<td>ACK F13</td>
</tr>
</tbody>
</table>
If a CANCEL crosses a 200 OK to an INVITE between Bob and a stateful proxy, the UAC may receive a 200 OK to the INVITE after receiving 200 OK to the CANCEL. TU must manage a CANCEL transaction and an INVITE transaction independently, and even if a CANCEL is successful, TU cannot terminate an INVITE transaction, as described on 9.1 of RFC3261. Like a CANCEL crossover of 2.1, the UAC may send a BYE and terminate the session immediately after receiving 200 OK to an INVITE.

Message Details

F1 INVITE Alice -> Proxy

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 INVITE Proxy -> Bob

INVITE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F3 100 Trying Proxy -> Alice

SIP/2.0 100 Trying
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Content-Length: 0

F4 180 Ringing Bob -> Proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.1
;received=192.0.2.233
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

F5 180 Ringing Proxy -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
F6 CANCEL Alice -> Proxy

CANCEL sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0

F7 CANCEL Proxy -> Bob

CANCEL sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.2
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0

/* Proxy sent a 200 OK to the CANCEL and Bob sent INVITE at the same time. */

F8 200 OK(INVITE) Bob -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.1
;received=192.0.2.233
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Content-Type: application/sdp
Content-Length: 148
v=0
c=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.100
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F9 200 OK(CANCEL) Proxy -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0

/* Since CANCEL requests are hop-by-hop, the proxy answers with a 200 OK to the CANCEL of Alice. Note that the 200 OK doesn’t mean the success of the CANCEL to the INVITE. */

F10 200 OK(INVITE) Proxy -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9 ;received=192.0.2.101
Record-Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 148

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

F11 481 Call/Transaction Does Not Exist Bob -> Proxy
SIP/2.0 481 Call/Transaction Does Not Exist
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.2
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 CANCEL
Content-Length: 0

/* Since Bob has already sent 200 OK to INVITE, CANCEL fails with 481 response. */

F12 ACK Alice -> Proxy
ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bq9
Max-Forwards: 70
Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

F13 ACK Proxy -> Bob
ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK721e.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bq9
;received=192.0.2.101
Max-Forwards: 69
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

F14 BYE Alice -> Proxy
BYE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74be5
Max-Forwards: 70
Route: <sip:ss.atlanta.example.com;lr>
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
/* Alice may send a BYE and terminate the session immediately on receipt of a 200 OK after the CANCEL. */

F15 BYE Proxy -> Bob

BYE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK739578.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74be5 ;received=192.0.2.101
Max-Forwards: 69
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

F16 200 OK Bob -> Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP ss.atlanta.example.com:5060;branch=z9hG4bK739578.1
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74be5 ;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

F17 200 OK Proxy -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74be5 ;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

2.10. A retransmitted 200 OK after sending a BYE

Alice   Bob
|---------------------->
| INVITE F1

Hasebe
In this scenario, a ACK request to a 200 OK response is lost (or delay), immediately after Bob sends the retransmitted 200 OK to ini-INVITE and Alice sends a BYE at the same time.

(Usually UAS don’t terminate the session with a BYE, but it’s possible in the case of a CANCEL crossover since Alice sends a BYE immediately after sending a ACK.)

Depending on the implement of a SIP user agent, Alice may start a session again by reception of the retransmitted 200 OK with SDP since she has already terminated a session by sending a BYE. In that case, if UAC receives a retransmitted 200 OK after sending a BYE, you should not start a session again since the session which is not associated with dialog remains. Moreover, in the case where UAS sends an offer with a 200 OK, if UAS receives a retransmitted ACK after receiving a BYE, UAS should not start a session again for the same reason.

Message Details

F1 INVITE Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0

F3 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 147

v=0
o=bob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
F4 ACK Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* A ACK request is lost. */

F5 200 OK Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 147

v=0
c=boob 2890844527 2890844527 IN IP4 client.biloxi.example.com
s=-
c=IN IP4 192.0.2.201
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/* UAS retransmits a 200 OK to an ini-INVITE since it didn’t receive
a ACK. */

F6 BYE Alice -> Bob

BYE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
/* Bob retransmits a 200 OK and Alice sends a BYE at the same time. */

F7 ACK Alice -> Bob

ACK sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds8
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 ACK
Content-Length: 0

/* UAC retransmits a ACK since it received a retransmited 200 OK. Alice has already terminated a session by sending a BYE. She should not start a session again by receiving a retransmited 200 OK. */

F8 200 OK(BYE) Bob -> Alice

SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnashds9 ;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0

/* Bob sends a 200 OK to a BYE. */

Editor’s Note
It would be needed clarify related with Dialogue status and the waiting timer of re-send message.

2.11. A BYE on the early dialog

<table>
<thead>
<tr>
<th>Alice</th>
<th>Bob</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE F1</td>
<td></td>
</tr>
<tr>
<td></td>
<td>--------------</td>
</tr>
<tr>
<td>180 Ringing F2</td>
<td></td>
</tr>
<tr>
<td>&lt;--------------</td>
<td></td>
</tr>
<tr>
<td>BYE F3</td>
<td>200 OK(BYE) F4</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>
In this scenario, Alice establishes an early dialog with the receiving 180 response. Alice sends a BYE on the early dialog. According to Section 15 of RFC3261, callee’s UA MUST NOT send a BYE on early dialogs, but the caller’s UA MAY send a BYE on early dialogs.

Message Details

F1 INVITE Alice -> Bob

INVITE sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Type: application/sdp
Content-Length: 151

v=0
o=alice 2890844526 2890844526 IN IP4 client.atlanta.example.com
s=-
c=IN IP4 192.0.2.101
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2 180 Ringing Bob -> Alice

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0
/* Alice forms an early dialog by receiving a 180 response to ini-INVITE. However Bob is not sure that Alice received the 180 response. */

F3 BYE Alice -> Bob

```
BYE sip:bob@client.biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bKnsashds9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

/* Alice sends a BYE on the early dialog and Alice terminates a session (if any). */

F4 200 OK(BYE) Bob -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 2 BYE
Content-Length: 0
```

/* Bob sends a 200 OK to a BYE of Alice, and Bob terminates a session (if any). */

F5 487 Request Terminated(INVITE) Bob -> Alice

```
SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bd5
;received=192.0.2.101
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>;tag=8321234356
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 INVITE
Contact: <sip:bob@client.biloxi.example.com;transport=udp>
Content-Length: 0
```

/* Bob should terminate the early dialog when he receives a BYE. Bob sends a 487 response to terminate a INVITE transaction in the similar way to handle a CANCEL from Alice, because the INVITE transaction remains after terminating the early dialog. */
F6 ACK Alice -> Bob

ACK sip:bob@biloxi.example.com SIP/2.0
Via: SIP/2.0/UDP client.atlanta.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:alice@atlanta.example.com>;tag=9fxced76sl
To: Bob <sip:bob@biloxi.example.com>
Call-ID: 3848276298220188511@atlanta.example.com
CSeq: 1 ACK
Contact: <sip:alice@client.atlanta.example.com;transport=udp>
Content-Length: 0

/* Alice sends an ACK to a 487 response as processing of the ini-INVITE transaction. (The dialog has been already terminated, but the ini-INVITE transaction remains) */

3. References


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